DBC 422 and DBC 425

INSTALLATION INSTRUCTIONS

AASTRA

1 General

These installation instructions are valid for the IP phones DBC 422, also called Dialog 4422 and for DBC 425, also called Dialog 4425.

These telephones use the H.323 protocol.

1.1 Scope

These IP phones can be connected to a number of Aastra's PBXes or equivalent.

Connected to MX-ONE TSW (ASB 501 04)

These IP phones are connected via a LAN to the IP device board IPLU/ELU32. The board together with the system software works as a gatekeeper, providing address translation, bandwidth management, admission control and call control.

A limited number of IP terminals can be registered towards each IPLU/ELU32 board. For capacity see the installation planning for *IP EXTENSION*.

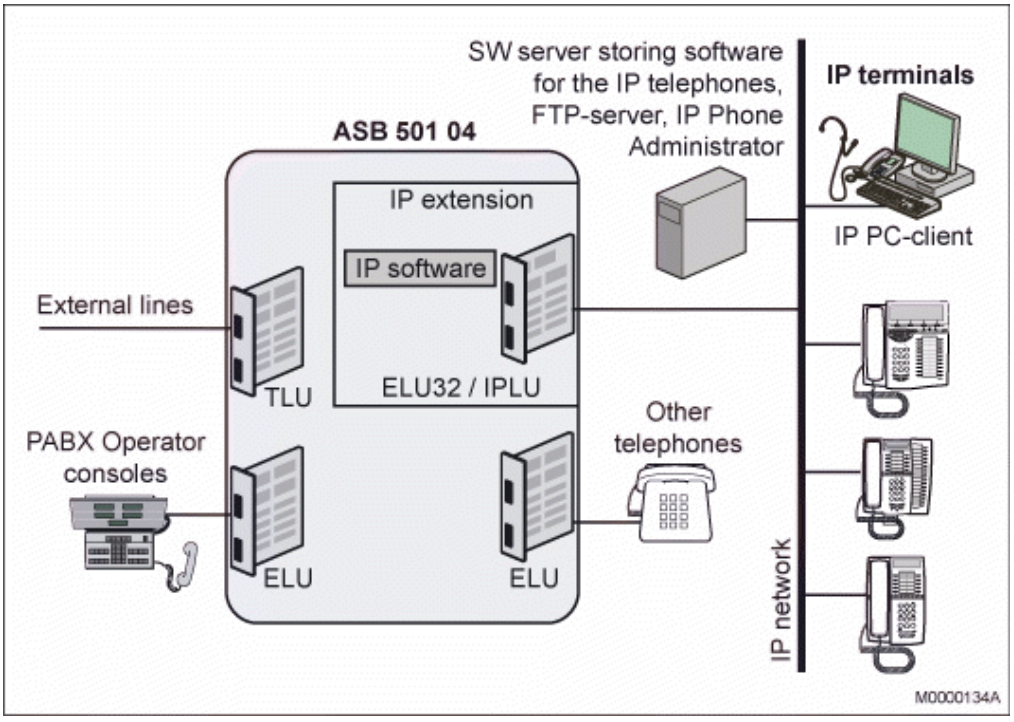


Figure 1: Connection of the IP phone towards MX-ONE TSW (ASB 501 04)

A password for each IP extension can be initiated in the exchange. The password is used to check that the user is allowed to log on with the entered extension number.

ASB 501 04 has support for automatic gatekeeper discovery, which means that the IP address to the gatekeeper (IPLU/ELU32 board) is retrieved automatically in the IP phone.

Connected to MX-ONE TSE

The figure below shows a typical setup for the MX-ONE Telephony Server.

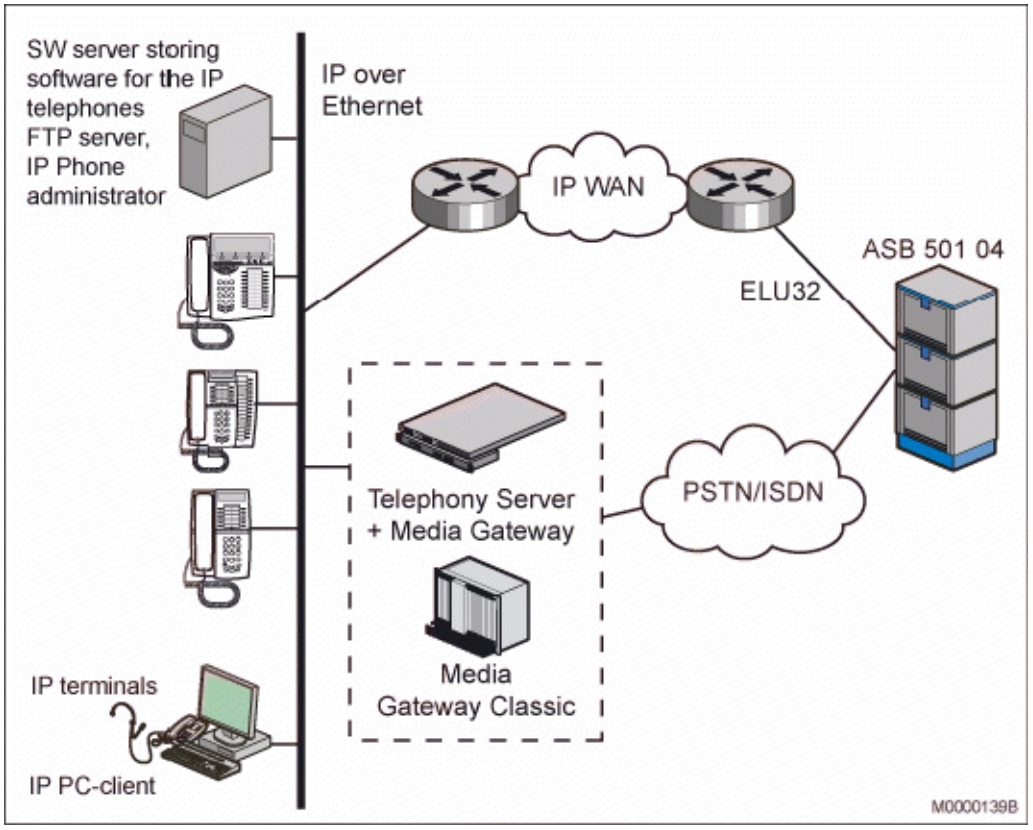


Figure 2: Typical MX-ONE Telephony Server setup

Connected to BusinessPhone

These IP phones can be used together with BusinessPhone version 7 or later.

Connected to MD Evolution

These IP phones can be used together with MD Evolution version 8 or later

1.2 Environmental requirements

The products covered in these installation instructions comply with the prerequisites stipulated for placing appliances in office and exchange room environments.

2 Aids

Wall mounting requires additional screws and spacers, 7.50 Wall mounting of the IP phone on page 88

3 Preparations

Check that an Ethernet cable is available and verify that it is possible to connect to the LAN.

4 Power equipment

The IP phone can either be powered from a 24 V AC/AC adapter or from a power hub. If it is powered from an adapter the following alternatives exist:

- RES 141 312/1 for the EU market except for the UK (230 V)
- RES 141 314/1 for the UK market (230 V)
- RES 141 318/1 for the 110 V markets

For other markets the AC/AC adapter is locally sourced. The phone can also be powered with 24-48 Volts DC.

Power consumption: 2.6 W (only the phone) and 4 W with the AC/AC adapter included. An extra key panel requires 0.03 W.

As an alternative the IP phone can be powered via the LAN from a power hub. The phone supports the standard IEEE 802.3AF for power over LAN.

These phones comply with the power class signature. The phone reports power class 1 which means less than 4 W required.

5 Earthing/grounding

No special earthing or grounding is needed.

The IP phone needs a shielded Ethernet cable for the network connection.

6 Cabling

The maximum line length between an IP phone and the LAN is 100 metres (328 feet) according to IEEE 802.3. Category 5 cables are recommended.

The following Ethernet category 5 cable can be ordered from Aastra:

- TSR 901 0452/3000

The figure shows where all the cables are to be connected in the bottom of the phone.

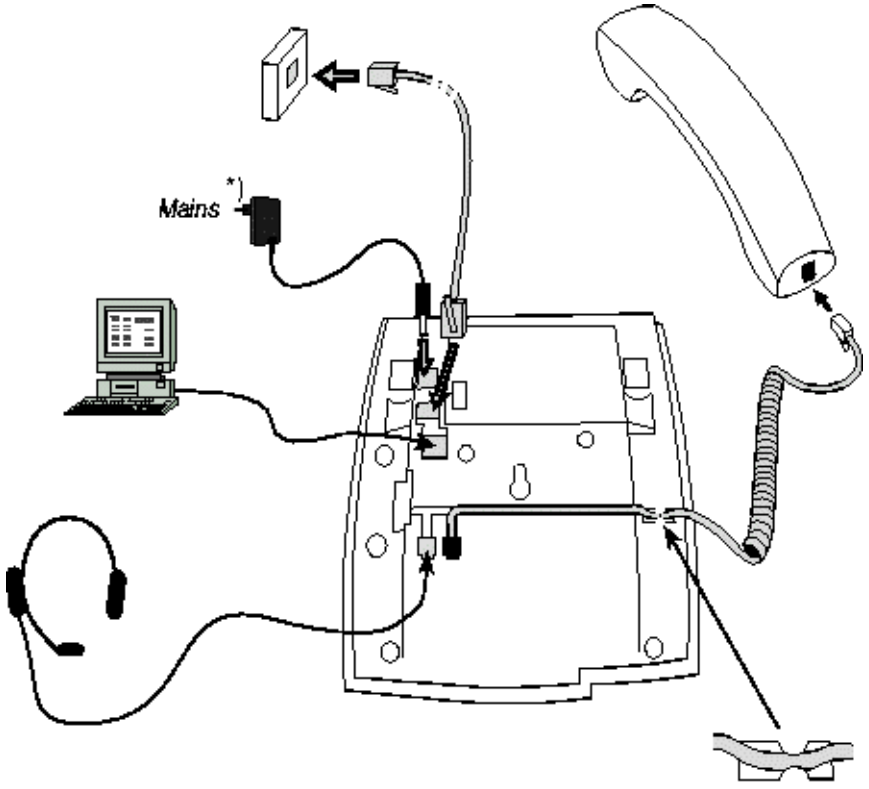


Figure 3: Connection of the phone

7 Installation

7.1 How to start a new phone

To select the protocol, 7.6 Select H.323 protocol on page 21.

Connect one end of the network cable to the network outlet and the other end to the connector marked LINE on the bottom of the IP phone. 6 Cabling on page 7.

If there is no LAN connection the text **No connection to the network** is shown in the display.

This section is divided into one part if a DHCP server is used, 7.3.1 Starting a DBC 425 phone in a LAN with a DHCP server on page 14 and into one part if fixed IP addresses are to be used 7.3.2 Manually setting of the fixed IP addresses on page 18.

7.2 Starting a DBC 422 phone

7.2.1 Starting a phone in a LAN with a DHCP server

This section describes the procedure when the phone will use IP addresses provided by a DHCP server.

After power up of the IP phone the LED at the mute key will be lit for a couple of seconds. Then the display will show:

Figure 4:

The **x** indicates a timer counting down seconds.

If the system administrator wants to change the IP settings, the administrator mode must be entered before the timer x is counted down to zero. If no procedure for the administrator mode is entered, or if the **Speaker** -key is pressed, the phone will use the current stored settings and continue with the menu LAN access control.

If the procedure for administrator mode is entered, the following menu appears:

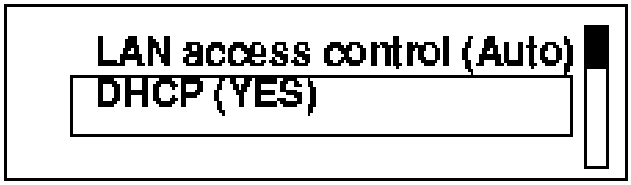


Figure 5:

Verify that the DHCP parameter is set to *Yes*, otherwise press the **Speaker** key and change the value to *Yes*.

Normally the phone gets the IP address to the SW server automatically from DHCP option 43 or from DNS SRV records. Step down in the **Network** menu list and check that the parameter for **Auto SW Server** is set to *Yes*.

A number of more IP settings can be done. For a complete survey of all menus in the Network settings, see 7.34.4 Change IP settings in DBC 422 on page 70.

Use the **C** -key to leave the settings menu and continue the boot sequence.

If LAN access control according to IEEE802.1x is enabled in the LAN, the following menu appears:

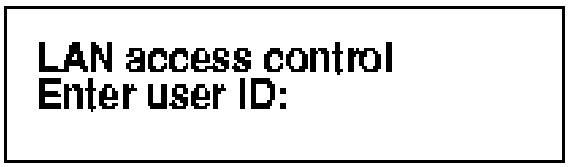


Figure 6:

Enter the LAN access control user identity and press the **Speaker** key.

Figure 7:

Enter the LAN access control password and press the **Speaker** key.

If the authentication is successful the normal boot sequence continues:

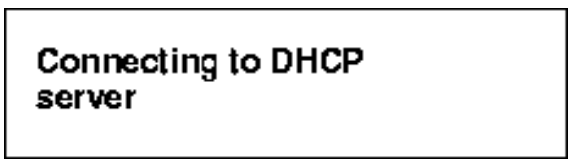


Figure 8:

The phone fetches the own IP address and other data from the DHCP server, see 7.11.2 Data from DHCP on page 30.

Then the following menu is shown:

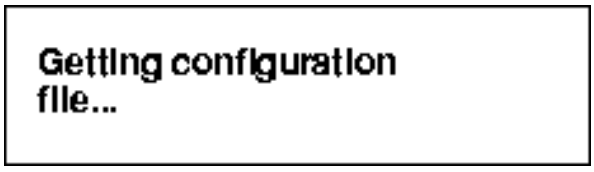


Figure 9:

The phone fetches the configuration file from the SW server. If the phone cannot get the file from the SW-server, it will use the one that is already stored in the local memory.

The phone checks for new software. If the software will be updated continue to 7.3.3 Update of the software in the IP phone on page 19.

Otherwise the following menus are shown:

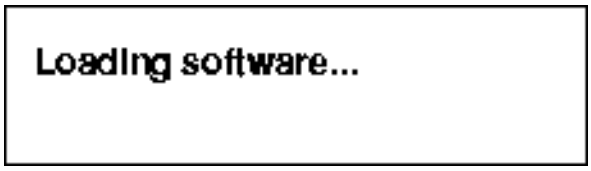


Figure 10:

The software is loaded internally in the phone. This process takes about 15 seconds.

Figure 11:

The phone performs a test to verify operation of the phone circuits.

The next step is to set the IP address to the gatekeeper. To set the IP address to the gatekeeper in H.323, see 7.14 Gatekeeper address on page 46.

If a password or Personal Identification Number (PIN) is to be used for this extension number, the password must be initiated in the exchange.

In the start up sequence for the phone, the next menu is:

Figure 12:

If the emergency call function is disabled the display shows:

Figure 13:

The next step is to register the phone towards the gatekeeper. The IP extension must already be initiated in the system. The directory number used at the previous log on is shown. If the number is to be changed, enter the new extension number. Press the **Speaker** key to log on.

If the gatekeeper requires that a password or PIN must be used, the following menu is displayed:

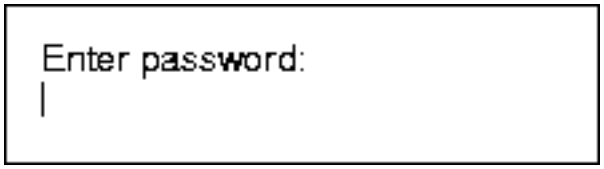


Figure 14:

Enter the password or PIN and press the **Speaker** key.

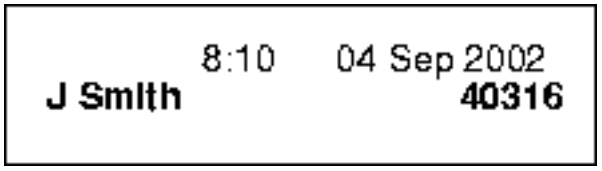


Figure 15:

Now the phone is ready for making and receiving calls.

The description of how to use the phone, see the directions for use for each platform.

7.2.2 Manually setting of the fixed IP addresses

If a DHCP server is not used in the LAN, fixed IP addresses must be used. The fixed IP addresses must be entered in the phone manually.

After power up of the IP phone, the display will show a special pattern and the headset LED will be lit for a couple of seconds. Then the display will show:

Figure 16:

The **x** indicates a timer counting down seconds.

To change the IP settings, enter the administrator mode before the timer x is counted down to zero. If the administrator mode is not entered, or C is pressed, the currently stored settings will be used and the *Getting configuration file* menu is shown.

If the procedure for administrator mode is entered, the following menu appears:

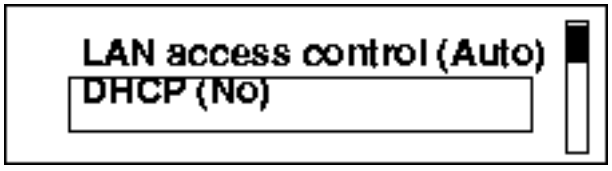


Figure 17:

If fixed IP addresses will be used, verify that the value for DHCP is *No*. To change this value, press the **Speaker** key, the **+** key and then press the **Speaker** key. The Network menu appears again. For a complete survey of all menus in the Network settings, see 7.34.4 Change IP settings in DBC 422 on page 70.

Enter the IP settings by using the +, - and Speaker keys for the entries in the list.

Press the **C** key to exit from the settings menu and to continue the boot procedure.

After this, the start up sequence is similar to the one when using DHCP, see 7.2.1 Starting a phone in a LAN with a DHCP server on page 8.

7.2.3 Update of the software in the IP phone

This section is reached from 7.2.1 Starting a phone in a LAN with a DHCP server on page 8 or from 7.2.2 Manually setting of the fixed IP addresses on page 11.

Two types of updates may be performed on the IP phone:

- Update of both the application software and the bootROM.
- Update of only the application software.

Note: It is not possible to only update the boot software. If the boot software will be updated, the application has to be updated as well.

The following menu appears if a new bootrom shall be loaded:

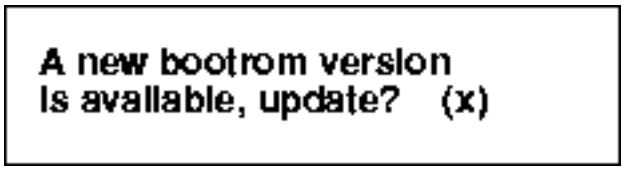


Figure 18:

Press the **Speaker** key to update the software.

The **x** indicates a timer counting down seconds. If the **C** key is not pressed during this time, the update is selected automatically.

If the **C** key is pressed, the phone will use the current version of the software.

When the **Speaker** key is pressed, the following menus are shown:

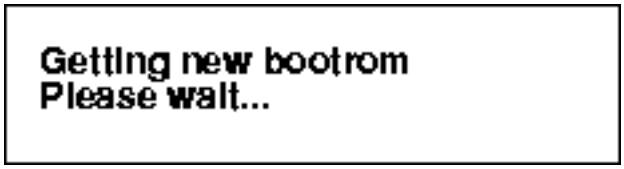


Figure 19:

When the bootrom firmware is loaded, the phone restarts and after a while the following menu is shown:

Figure 20:

The application firmware is fetched from the SW server and loaded into the IP phone and then the following menu is displayed:

Figure 21:

It is important that the power to the phone is not disconnected while this text is shown in the display. This process takes a couple of minutes. If

there is a power failure during this phase, the phone has to load the software again.

The next menu shows **Selftest OK.**

The next step is to set the gatekeeper's IP address and register the extension towards the gatekeeper. This is described in 7.2.1 Starting a phone in a LAN with a DHCP server on page 8.

7.3 Starting a DBC 425 phone

7.3.1 Starting a DBC 425 phone in a LAN with a DHCP server

This section describes the procedure when the phone uses IP addresses provided by a DHCP server.

After power up of the IP phone the mute key LED will be lit for a couple of seconds. Then the display will show:

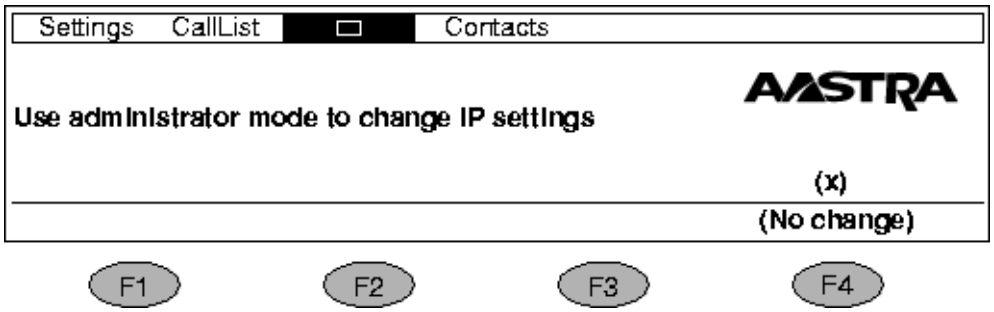


Figure 22:

The **x** indicates a timer counting down seconds.

If the installation personnel want to change the IP settings, the administrator mode must be entered before the timer x is counted down to zero. If no procedure for the administrator mode is entered, or if **No change** (F4) is pressed, the phone will use the current stored settings and continue with the menu LAN access control.

If the procedure for administrator mode is entered, the following menu appears:

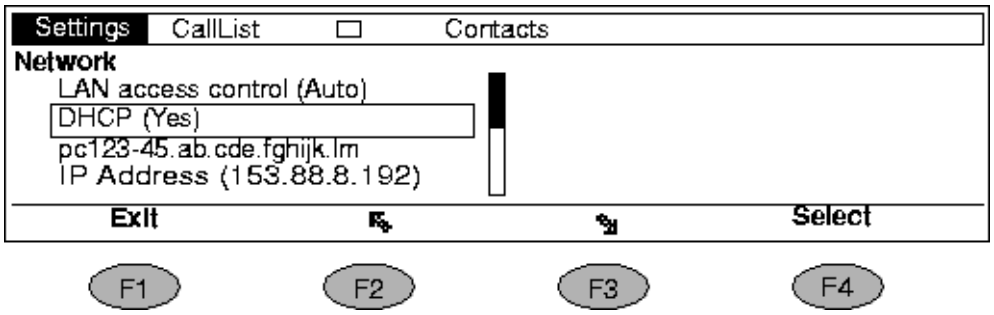


Figure 23:

Verify that the DHCP parameter is set to *Yes*, otherwise press the **Select** (F4) key and change the value to *Yes*.

Normally the phone gets the IP address to the SW server automatically from DHCP option 43 or from DNS SRV records. Step down in the **Network** menu list and check that the parameter for **Automatic SW Server** is set to *Yes.*

A number of more IP settings can be done. For a complete survey of all menus in the Network settings, 7.34.5 Change IP settings in DBC 425 on page 73.

Use the home key to leave the settings menu and continue the boot sequence.

If LAN access control according to IEEE802.1x is enabled in the LAN, the following menu appears

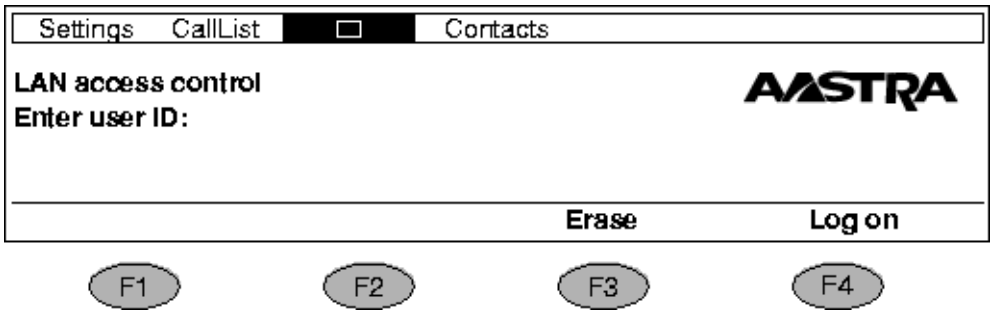


Figure 24:

Enter the LAN access control user identity and press the log on soft key.

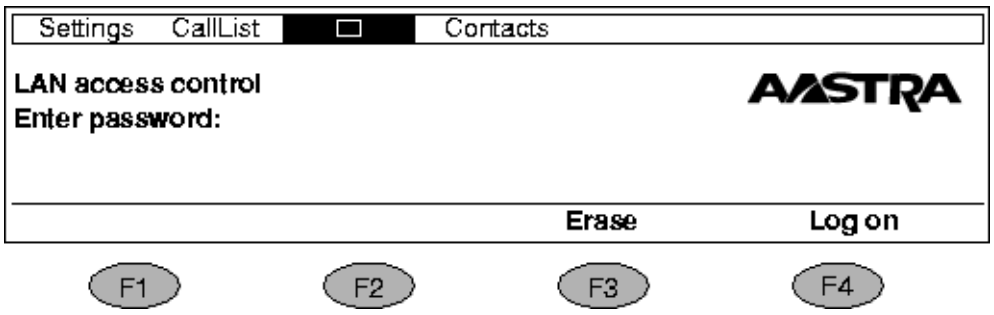


Figure 25:

Enter the LAN access password and press the **Log on** key.

If the authentication is successful the normal boot sequence continues. The following menu appears:

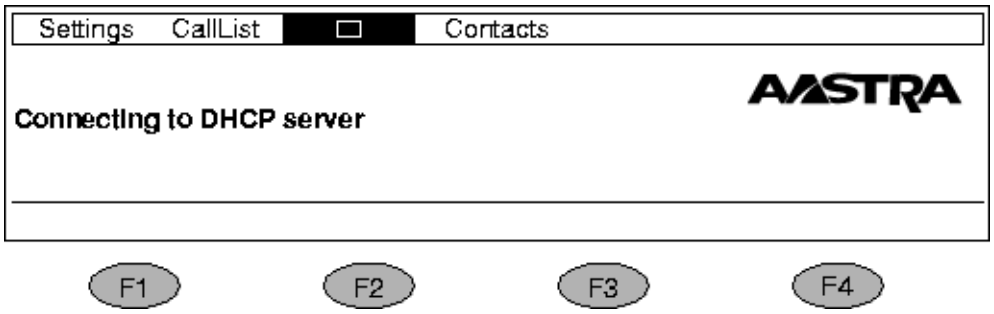


Figure 26:

The phone fetches the own IP address and other data from the DHCP server, 7.11.2 Data from DHCP on page 30.

Then the following menu is shown:

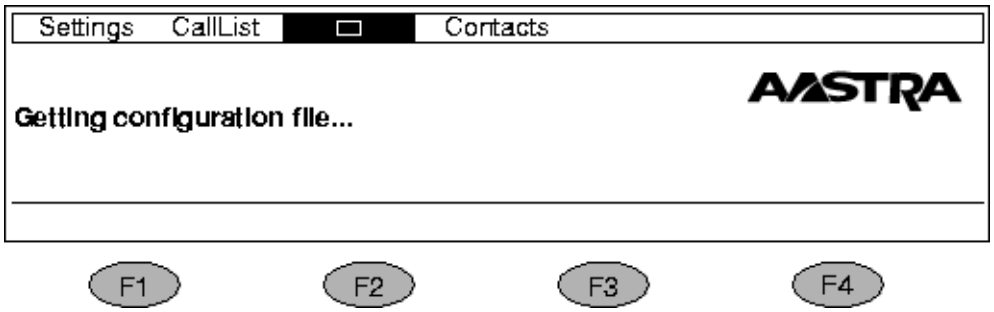


Figure 27:

The phone tries to fetch the configuration file from the SW server. If the phone cannot get the file from the SW-server, it will use the one that is already stored in the local memory.

The phone checks for new software. If the software will be updated continue to section 7.3.3 Update of the software in the IP phone on page 19.

Otherwise the following menus are shown:

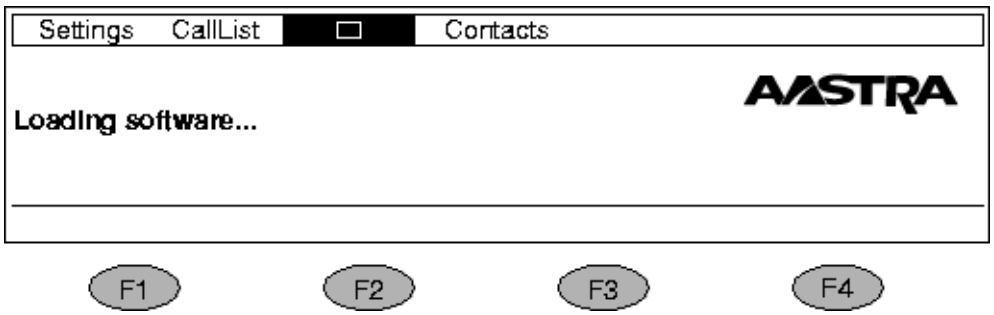


Figure 28:

The software is loaded internally in the phone. This process takes about 15 seconds.

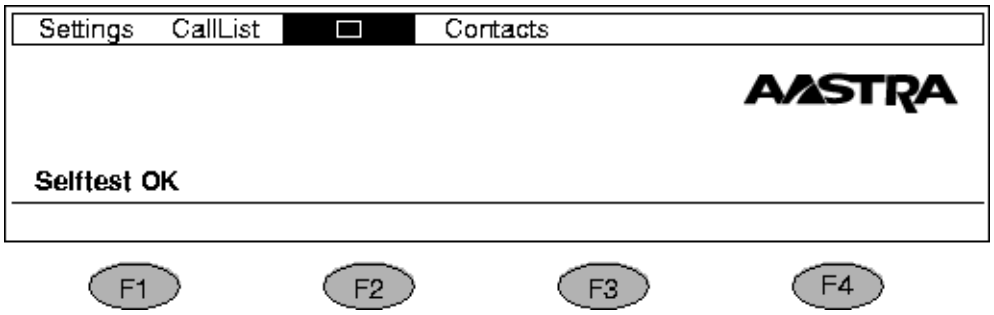


Figure 29:

The phone performs a test to verify operation of the phone circuits.

The next step is to set the IP address to the gatekeeper. To set the IP address to the gatekeeper in H.323, 7.14 Gatekeeper address on page 46.

If a password or Personal Identification Number (PIN) is to be used for this extension number, the password or the PIN must be initiated in the exchange.

In the startup sequence for the phone, the next menu is:

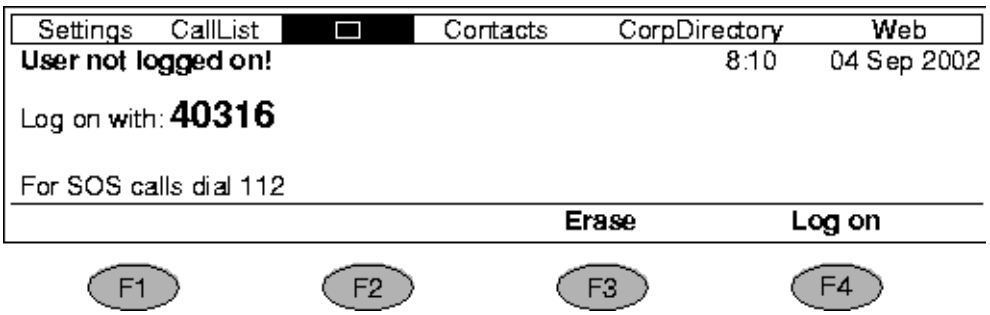


Figure 30:

The next step is to register the phone towards the gatekeeper. The IP extension must already be initiated in the system. The directory number used at the previous log on is shown. If the number must be changed, enter the new extension number. Press **Log on** (F4).

If the emergency call function is disabled the SOS text is not shown. To enable, 7.38 Emergency call on page 76.

If the gatekeeper requires that a password or PIN must be used, the following menu is displayed:

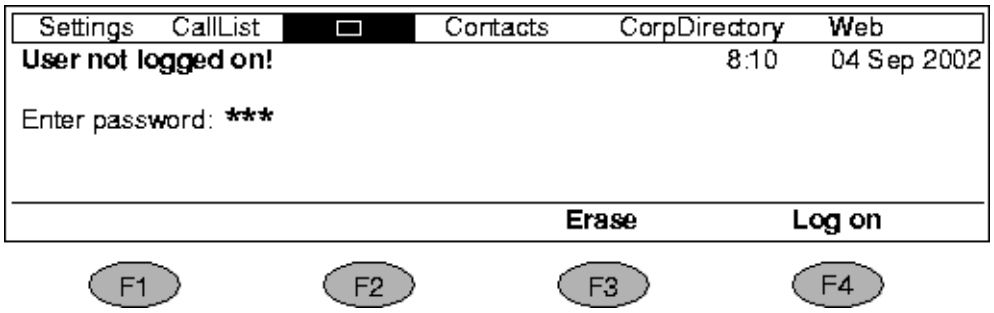


Figure 31:

Enter the password or PIN and press **Log on** (F4).

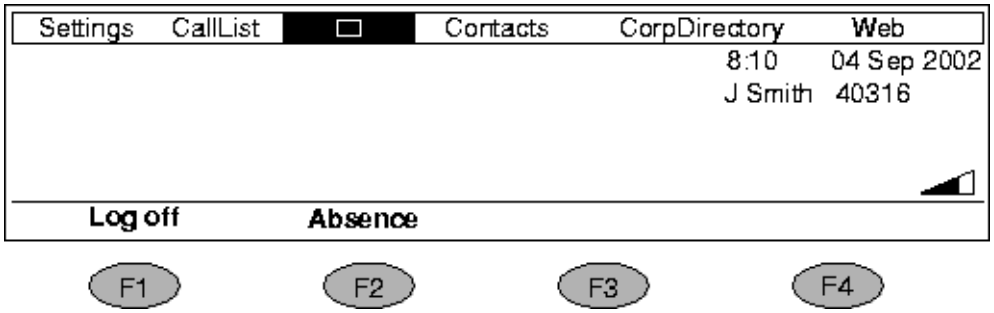


Figure 32:

Now the phone is ready for making and receiving calls.

For a description of how to use the phone, see directions for use for each platform.

7.3.2 Manually setting of the fixed IP addresses

If a DHCP server is not used in the LAN, fixed IP addresses will be used. The fixed IP addresses must be entered in the phone manually.

After power up of the IP phone the mute key LED will be lit for a couple of seconds. Then the display will show:

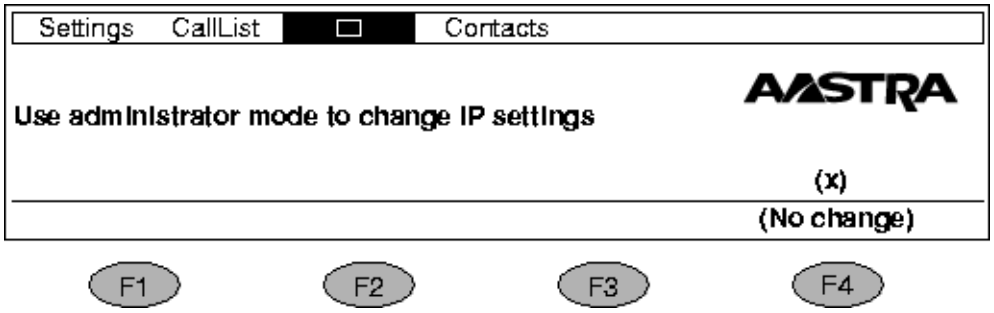


Figure 33:

The **x** indicates a timer counting down seconds.

To change the IP settings, enter the administrator mode before the timer x is counted down to zero. If the administrator mode is not entered, or **No change** (F4) is pressed, the currently stored settings will be used and the *Getting configuration file* menu is shown.

If the procedure for administrator mode is entered, the following menu appears:

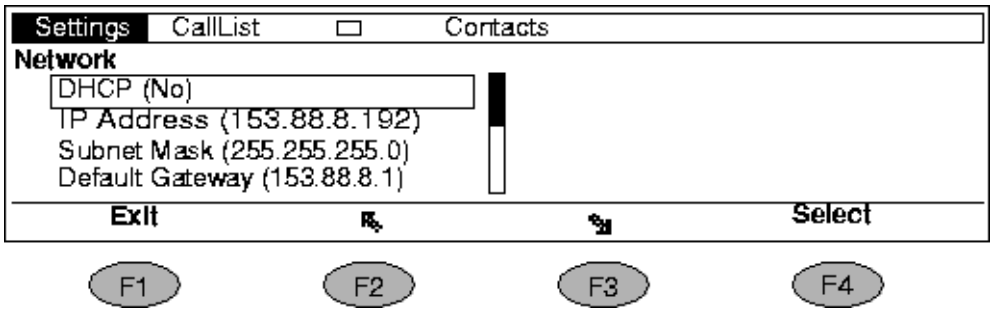


Figure 34:

If fixed IP addresses will be used, verify that the value for DHCP is No. For a complete survey of all menus in the Network settings, 7.34.5 Change IP settings in DBC 425 on page 73.

Enter the necessary IP settings.

Press **Exit** (F1) to continue the boot procedure.

After this, the start up sequence is similar to the one when using DHCP, see 7.3.1 Starting a DBC 425 phone in a LAN with a DHCP server on page 14.

7.3.3 Update of the software in the IP phone

This section is reached from the section, 7.3.1 Starting a DBC 425 phone in a LAN with a DHCP server on page 14 or from the section 7.3.2 Manually setting of the fixed IP addresses on page 18.

Two types of updates may be performed on the IP phone:

- Update of both the application software and the bootROM.
- Update of only the application software.

Note: It is not possible to only update the boot software. If the boot software will be updated, the application has to be updated as well.

The following menu appears if a new bootrom shall be loaded:

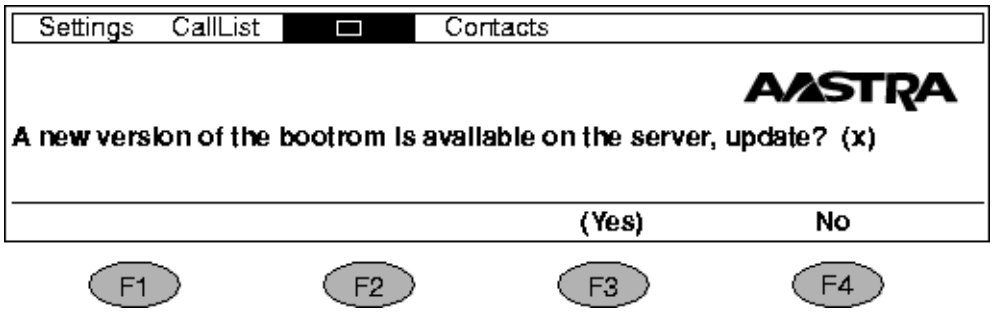


Figure 35:

Press **(Yes)** (F3) to update the software.

The **x** indicates a timer counting down seconds. If the **No** (F4) key is not pressed during this time, **Yes** is selected automatically.

If **No** (F4) is pressed, the phone will use the current version of the software.

When **Yes** (F3) is pressed, the following menus are shown:

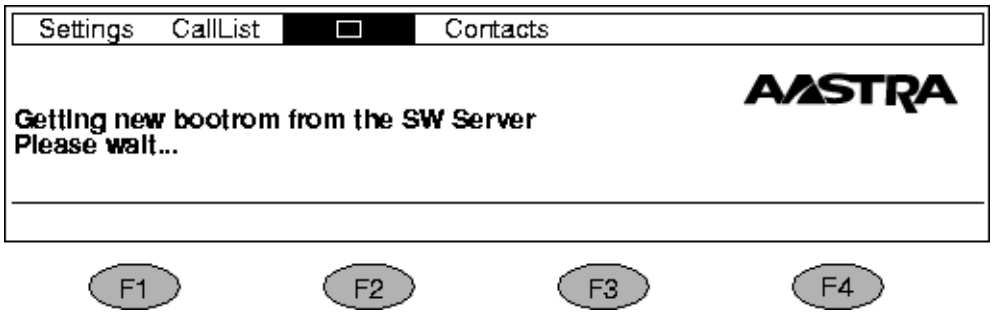


Figure 36:

When the bootrom firmware is loaded, the phone restarts and after a while the following menu is shown:

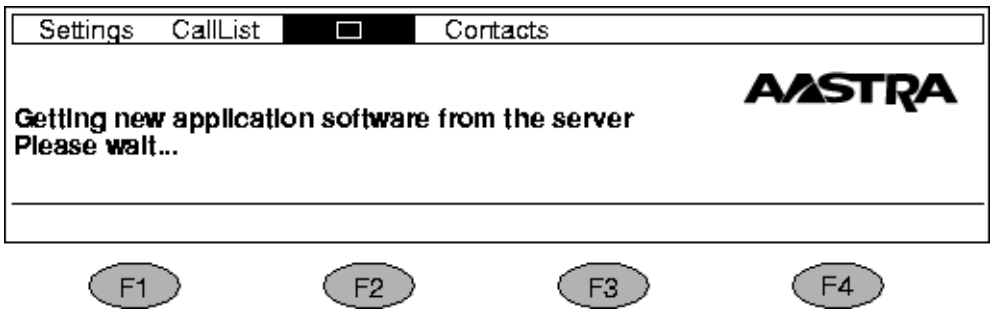


Figure 37:

The Application firmware is fetched from the SW server and loaded into the IP phone. This may take about one minute and the following menu is displayed:

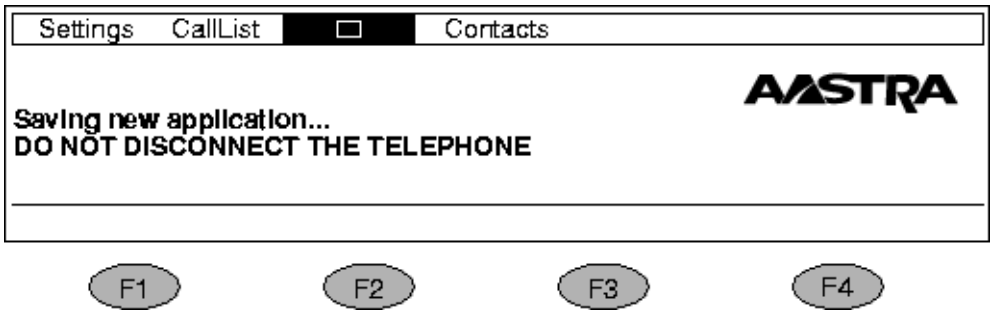


Figure 38:

It is important that the power to the phone is not disconnected while this text is shown in the display. This process takes a couple of minutes. If there is a power failure during this phase, the phone has to load the software again.

The next menu shows **Selftest OK**.

The next step is to set the gatekeeper's IP address and register the extension towards the gatekeeper. This is described in section 7.3.1 Starting a DBC 425 phone in a LAN with a DHCP server on page 14.

7.4 Delivery method

The IP phone is delivered in a box together with two foot consoles, one handset, one handset cord, designation labels, designation covers and an assembly instruction.

Extra key panels are delivered separately.

The phone is delivered with the software version that was valid when the phone was produced. The configuration file must be adapted for each site and has to be loaded into the phone, 7.7 SW loading on page 22.

7.5 Connection of the handset

The handset cord is connected with one end (short uncoiled) to the handset and the other end (long uncoiled) to the connector on the bottom of the IP phone marked HANDSET.

7.6 Select H.323 protocol

The phone is delivered with the H.323 protocol enabled.

7.7 SW loading

The software to be loaded into the phone is to be stored on a web server with HTTP protocol. This web server is called SW server in the menus and in this document. The following files are stored on the SW server:

d42x02-applic_R1A.dat
(CAA 158 0043) The application firmware for the DBC 42x 02 phones. R1A in the file name is an example.

d42x02-boot_R1A.dat
(CAA 158 0044) The boot ROM firmware. This software is used to be able to load the application into the IP phone. R1A in the file name is an example.

d42x02-config.txt
(CAA 158 0042) The configuration file. This file contains information about the version of the software to be used and other configuration data. Normally the configuration file has to be adapted for each installation, see the description for *CONFIGURATION FILE FOR DBC 42X*.

d42x02-lang_R1A.txt
(CAA 158 0045) The language file containing all the languages that are supported. R1A in the file name is an example. See the description for *LANGUAGE FILE FOR DBC42X 02*.

When the IP phone is powered up, the phone fetches the configuration file from the SW- server. If the software version defined in the configuration file is different than the software version in the phone, the phone fetches the application software file and/or the boot ROM software file from the SW server. The new software is automatically stored into the flash memory in the phone.

It is possible to load both newer and previous software versions with this method.

To check the software version in the phone, see 7.25 Software version on page 63

7.8 Several configuration files

A certain group of IP phones can often have different characteristics compared to the other groups of extensions concerning which codec to use, domain names, emergency number data etc. The following methods exist to get different configuration files for the groups of phones:

- Use the DNS (Domain Name Service) domain name received from DHCP, 7.11.2 Data from DHCP on page 30.

- Use the telephony domain name received in the vendor specific field in the DHCP messages, 7.11.2 Data from DHCP on page 30.
- Subnet method, 7.8.3 Subnet method on page 23
- Set the IP address of the SW server manually in the phone. In this case there must be one SW server per configuration file.

For all the methods, the corresponding directory names have to be created in the software server and the corresponding configuration files have to be stored under these directories.

7.8.1 DNS domain name

The DNS domain name, provided in option 15 in DHCP, is used to create the URI (universal resource identifier) to fetch the configuration file from the software server.
Example: /dns_domain_name/dbc42x02/d42x02-config.txt, see 7.9.3 Directory structure on page 25.

7.8.2 Telephony domain name

If the DNS domain name cannot be used, it is possible to create telephony domain names and these are sent as a tag in option 43 in DHCP. If the IP phone finds this tag, it will create the URI containing this domain name and fetch the configuration file from the software server. Example: /telephony_domain_name/dbc42x02/d42x02-config.txt, see 7.9.3 Directory structure on page 25.

7.8.3 Subnet method

The URI consists of the network address together with the subnet mask length. The network address consists of the IP address of the phone with a logical AND operation of the subnet mask.

Example: The phone has the IP address 130.100.26.144 and the subnet mask is 255.255.255.192. The AND operation gives the URI /130.100.26.128-26/dbc42x02/d42x02-config.txt. The component -26 is the length of the subnet mask (number of ones in the binary value of the subnet mask), see 7.9.3 Directory structure on page 25.

7.8.4 Priority between the different methods

The priority is:

- The telephony domain tag in option 43
- The DNS domain in option 15

- Subnet method
- The default configuration file is fetched. This file is stored under /dbc42x02/dbc42x02-config.txt.

7.9 Software Server (SW server)

A software server with the HTTP protocol is used for storing the firmware for the IP phone.

The IP address to the software server can be provided by one of the following methods:

- Manually in the **Network** menu or via the administrator web interface.
- DHCP, 7.11.2 Data from DHCP on page 30. This method has priority over the DNS SRV method.
- DNS SRV resource records, 7.10 DNS SRV resource records on page 27.

In a MX-ONE environment, the host for the Telephony Server and the IP phone software server cannot be the same.

7.9.1 Installation

Installation of the HTTP server should be done according to the manufacturer's documentation. Both PC and Unix versions are supported.

7.9.2 HTTP servers

As the SW server, the following http servers have been tested with the IP phone:

- Microsoft® NT4.
- Microsoft® Windows® 2000 and 2003 server. When using Windows® server the file type **.dat** must be enabled: In **IIS Manager** , go to **DefaultWEB Site** , select **Properties** , edit **HTTP header** , set **Associated extension: .dat** and set **Content type (MIME): application /octet-stream**
- Apache 1.3.3 on Microsoft® Windows® or on Redhat® Linux 5.2.
- Apache Tomcat. When the IP Phone Configuration File task in MX-ONE Manager TS shall be used the Tomcat server is mandatory. For more information, see the description for *CONFIGURATION FILE FOR DBC 42X*

The files according to section, see 7.7 SW loading on page 22, must be stored on the server. The file names must be according to what is described below, 7.9.3 Directory structure on page 25.

Note: When storing the files on the software server, make sure that the files are transferred in binary mode, otherwise extra bytes can be modified by the transfer tool and the size be changed. In this case the telephone will not load the file.

7.9.3 Directory structure

The directory structure under the http root directory must be created, see figure 39 Directory structure using domain names on page 26. When several different configuration files must be used for different groups of phones where each group is a member of a specific domain, the structure with different domain names are used. In this case the configuration files have the same name although they have different contents to define characteristics for the different groups of phones.

The domain name is described above, see 7.8 Several configuration files on page 22.

It is only the configuration file, and not the application and boot, that needs to be stored under each domain directory name.

If the phones do not find any configuration file in a domain directory, the file in the directory **web-server root/dbc42x02** is used.

The application, boot and language files are stored in the **dbc42x02** directory.

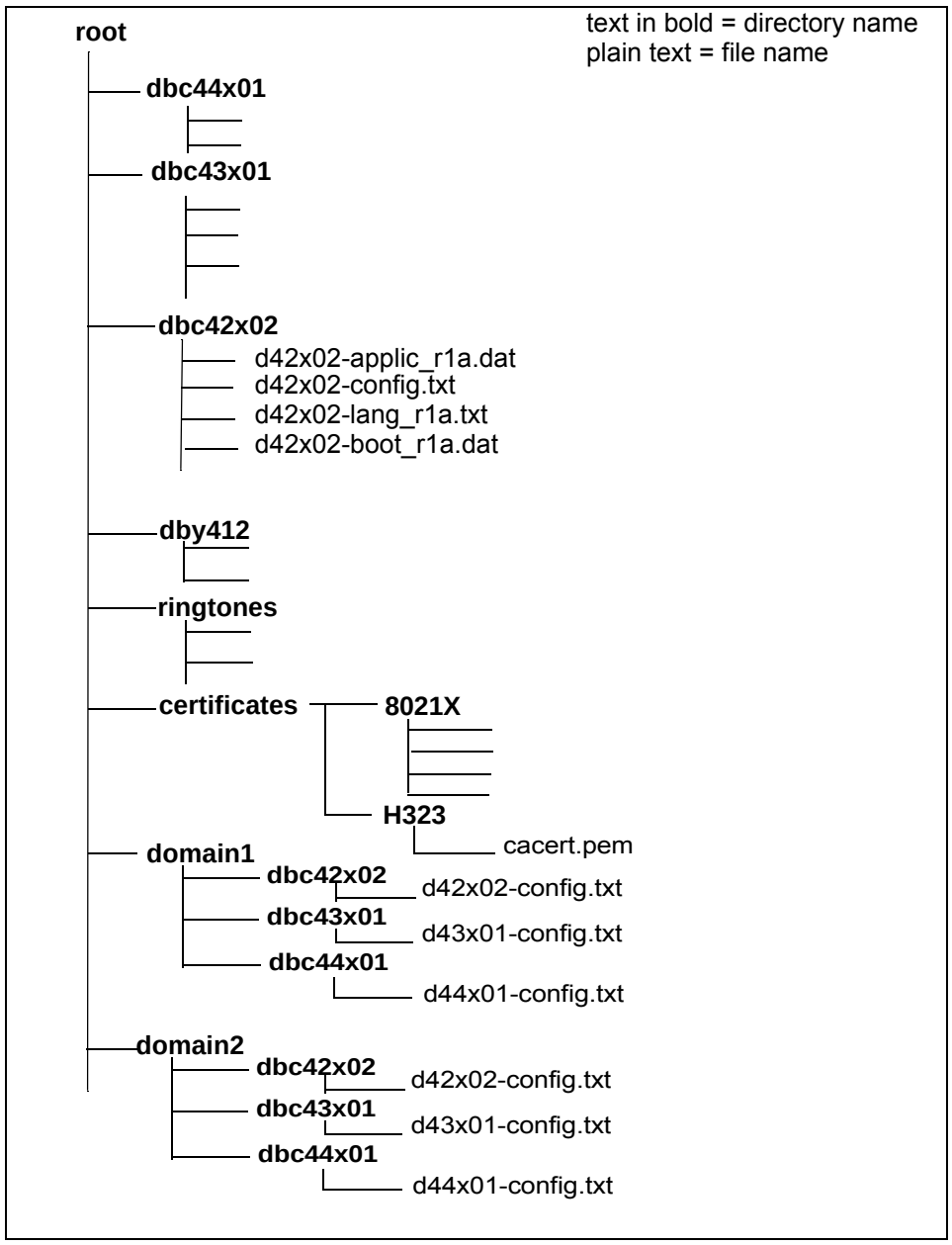


Figure 39:Directory structure using domain names

If the subnet method is used, 7.8.3 Subnet method on page 23, the directory structure will be as in the example below. In this example the phones belonging to the first group have the network address 130.100.26.128 with the subnet mask 255.255.255.192. The second group has the network address 130.100.27.0 with the subnet mask 255.255.255.0.

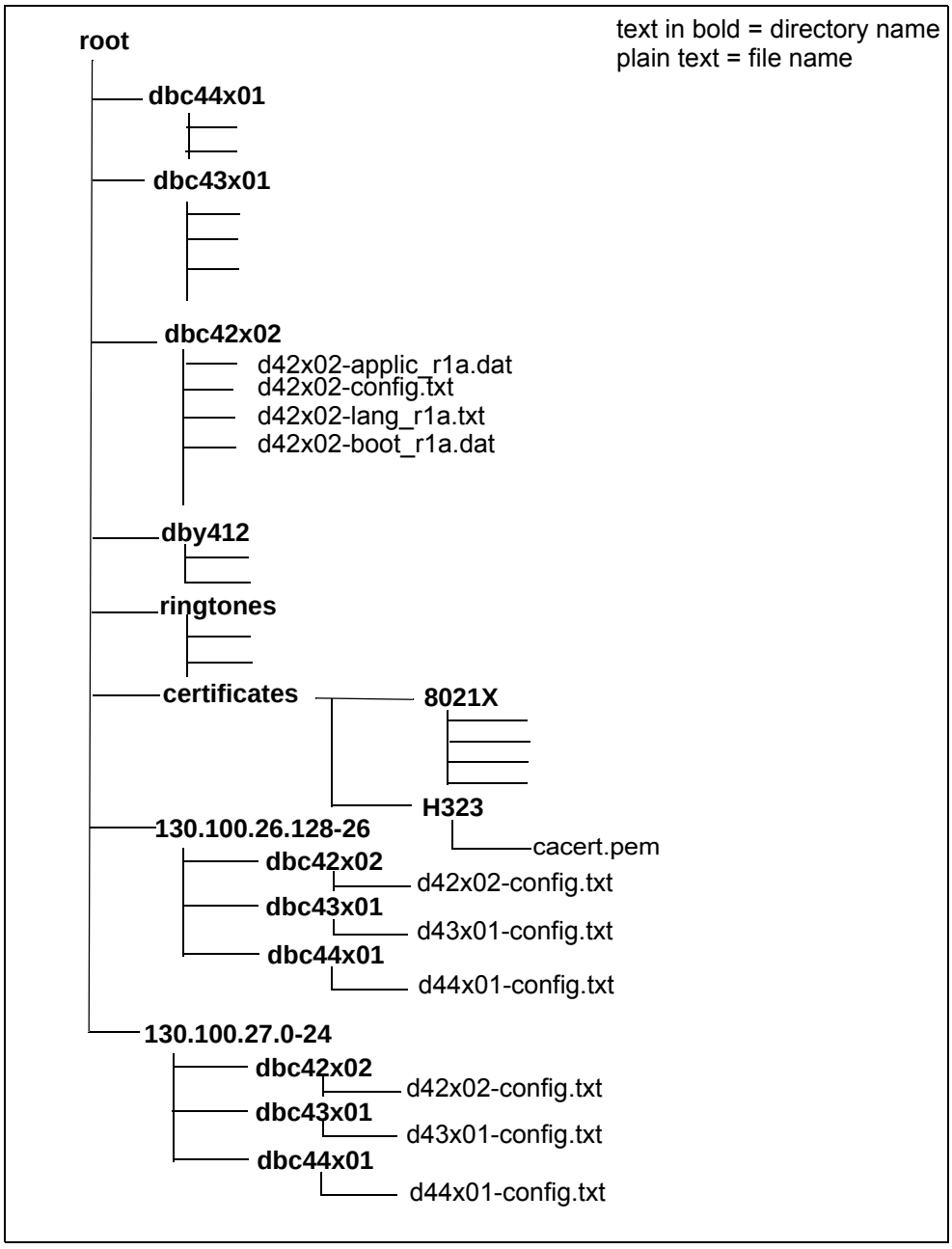


Figure 40:Directory structure using the subnet method

7.10 DNS SRV resource records

To get necessary IP addresses into the phone, one option is to use the DNS (Domain Name Server) SRV (service) resource records. The following data can be retrieved in this way:

- The IP address to the software server. To get this option, set the data in following way: In **Network** settings, chose **Automatic SW server = YES** but do not initiate this data in option 43 in DHCP. In

this case the phone will get the SW server IP address from DNS SRV.

- The IP address to the IP Phone Administrator server.
- The IP address to the SIP proxy server

The DNS SRV handling does only work when DHCP is used and when the DHCP server points out the DNS server and when a domain name is received in DHCP option 15. This service is described in the RFC 2782.

With this method the phone sends a request to the DNS server to get a particular service. This is an advantage compared to using option 43 in the DHCP messages, which all the devices on the LAN receive. If a device does not handle option 43 in a correct way, this can cause problems for this device.

In the answer from the DNS server the phone can get a list with hosts.

7.10.1 Enter data in DNS SRV resource records

The IP address to the software server

In the DNS SRV resource records, the following data has to be set to find the IP address to the SW-server:

Service

_aas442x_cfg._tcp (applications up to and including R7K)

_aasdbc_cfg._tcp (applications from R7M and later)

Protocol

_tcp

Prio

The priority of the target host. The phone will try to contact the target host with the lowest-numbered priority. Target hosts with the same priority should be tried in pseudo random order. The range is 0-65535.

Weight

A load balancing mechanism. When selecting a target host among those that have the same priority, the chance of trying this one first is proportional to its weight. The range is 1-65535. Domain administrators shall use Weight = 0 when there is not any load balancing to do.

Port

80 (fixed value)

Host

The DNS name of the Software server

The IP address to the IP Phone Administrator server

In the DNS SRV resource records, the following data has to be set to find the IP address to the IP Phone Administrator server:

Service

_aas442x_smgt._tcp (applications up to and including R7K)

_aasdbc_sgmt._tcp (applications from R7M and later)

Protocol

_tcp

Prio

The priority of the target host. The phone will try to contact the target host with the lowest-numbered priority. Target hosts with the same priority should be tried in pseudo random order. The range is 0-65535.

Weight

A load balancing mechanism. When selecting a target host among those that have the same priority, the chance of trying this one first is proportional to its weight. The range is 1-65535. Domain administrators shall use Weight = 0 when there is not any load balancing to do.

Port

8080

Host

The DNS name of the IP Phone Administrator server

7.10.2 Verification of entered data

The data entered into a DNS SRV resource record can be verified in a PC by:

- Open a DOS prompt window
- Enter the command **nslookup.** The response will show the current DNS server
- Enter **set type=srv**
- Enter the wanted service. Example: **_aasdbc_cfg._tcp.** domain name where the domain name is the one the phone receives from DHCP.
- The response will contain the DNS SRV resource record data, including the host name to the requested service

7.11 DHCP server

7.11.1 Installation

Installation of the DHCP (Dynamic Host Configuration Protocol) server should be done according to the documentation of the manufacturer. Both PC and Unix versions are supported.

The following DHCP servers have been tested with the IP telephone:

- Microsoft® Windows® NT4.
- Microsoft® Windows® 2000 and 2003 server.
- Redhat® Linux.

7.11.2 Data from DHCP

The telephone has support for DHCP by which the following IP configuration data can be provided:

- Own IP address, subnet mask and default gateway, received in the DHCP standard fields (1 and 3).
- The domain name for the LAN segment (DNS domain name) in code 15. The domain name is used in the automatic gatekeeper discovery routine, see section 7.15 Automatic Gatekeeper Discovery on page 47. It can also be used when several configuration files are used, see section 7.8 Several configuration files on page 22.
- The vendor specific field 43 can be used to get the following data:
  - IP address of the software server, see section 7.9 Software Server (SW server) on page 24.
  - IP address and port number of the http proxy server. If the software is to be loaded from a SW server outside the firewall the proxy settings are needed.
  - The telephony domain name. This can be used in the automatic gatekeeper discovery routine, see section 7.15 Automatic Gatekeeper Discovery on page 47. It can also be used when several configuration files are used, see section 7.8 Several configuration files on page 22.
  - A list with VLAN identities. These are used when the telephone will automatically be assigned to a VLAN, see section 7.20 Virtual LAN (VLAN) on page 51.
- DNS identity (web address) for the telephone.

For the complete usage of the domain name, see section 7.17 Domain name on page 49.

7.11.3 DHCP Settings for Option 43 and 60

DHCP option 60 (vendor class identifier) and option 43 (vendor specific information field) are used by the telephone to get the specific configuration data from the DHCP server. The flow is as follows:

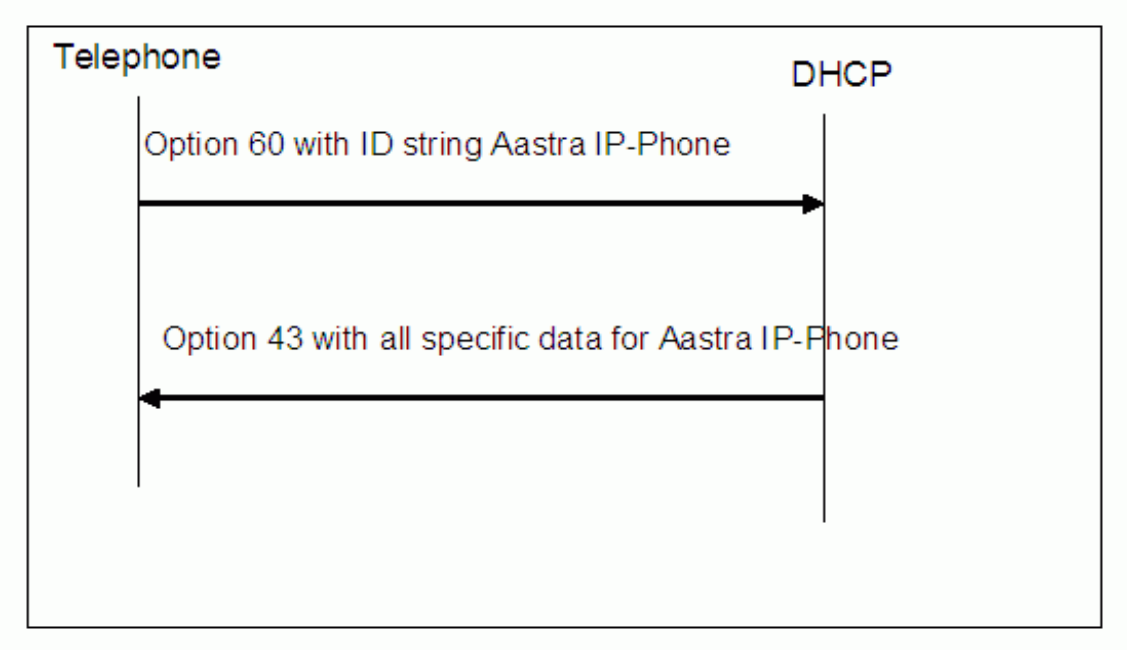


Figure 41:

The procedure to initiate the data for option 43 and 60 in the DHCP server is as follows:

1) define vendor class (option 60)
2) set predefines options (option 43)
3) set scope options (option 43)

7.11.3.1 Vendor Class Identifier

Vendor class identifier (option 60) option is used to secure that option 43 data for the specific vendor is sent from the DHCP server to the client. The telephone sends the vendor class identifier to the DHCP server, which returns vendor specific information for the requested vendor class in option 43 to the telephone.

When vendor class identifier shall be used to get the option 43 data for the Aastra IP-Phone, it is necessary to initiate the vendor class *Aastra IP-Phone* in the DHCP server and in some cases also the vendor class

Ericsson IP-Phone, see section 7.11.3.2 Vendor Specific Information Field on page 32.

7.11.3.2 Vendor Specific Information Field

The vendor specific information field (option 43) is coded as shown in the figure below.

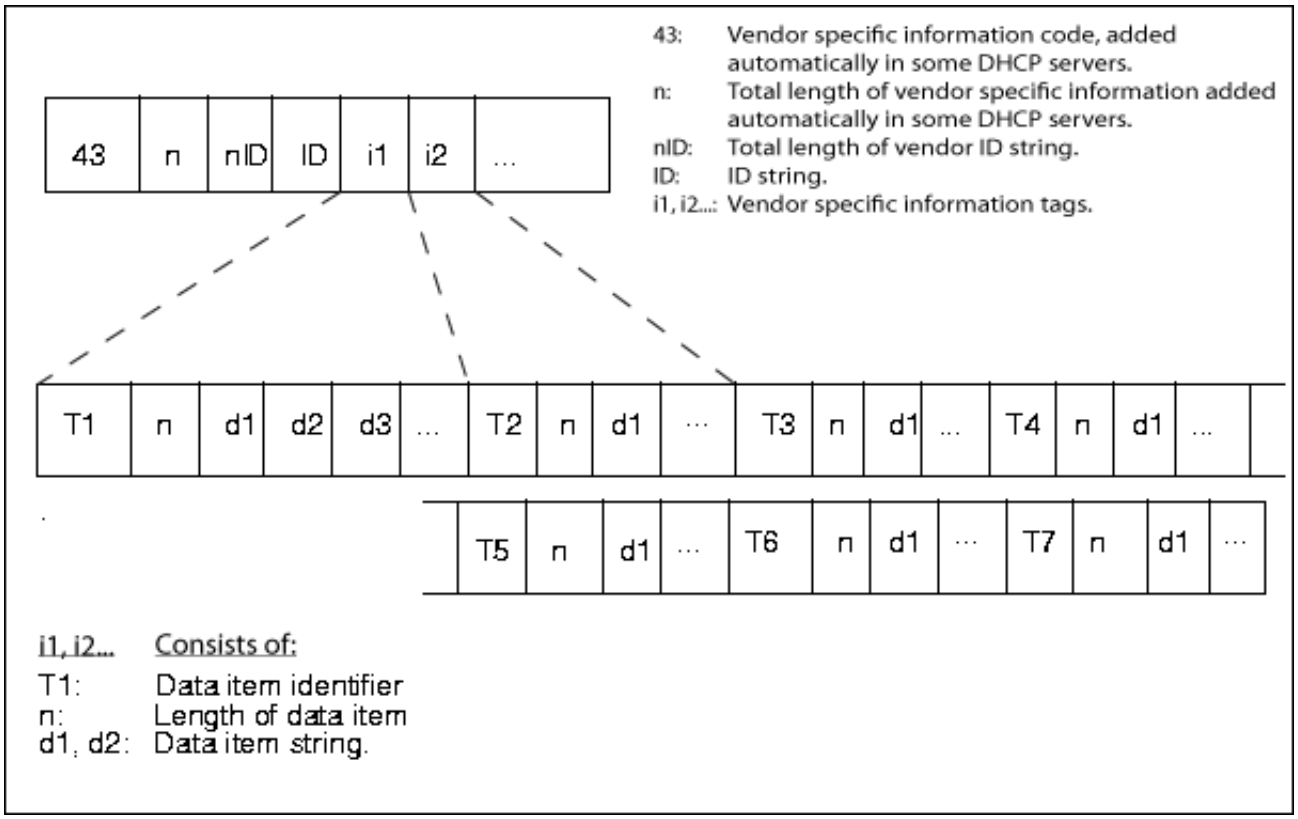


Figure 42:Vendor Specific Information structure

Within this vendor field, a substructure is used with the different tags:

Tag 01 (T1 in the figure): SW server's IP address in ASCII text format.

Tag 02 (T2 in the figure): Proxy server's IP address also in ASCII text format.

Tag 03 (T3 in the figure): Proxy port, also this in ASCII text format.

Tag 04 (T4 in the figure): Telephony domain name in ASCII text format.

Tag 05 (T5 in the figure): VLAN identity 1 for the telephone, in ASCII text format.

Tag 06 (T6 in the figure): VLAN identity 2 for the telephone, in ASCII text format.

Tag 07 (T7 in the figure): VLAN identity 3 for the telephone, in ASCII text format.

Note: The VLAN identity for the telephone defined here in option 43 must not be equal to the VLAN identity for the PC defined in the configuration file.

For more details about VLAN identity, see section 7.20 Virtual LAN (VLAN) on page 51.

The different tags are optional, but if tag 02 is used tag 03 is mandatory.

The following applies for the ID string:

- At new installation the string *Aastra IP-Phone* shall be entered in DHCP option 43.
- At upgrading (to application R7K or later and boot R3S or later) of a site where the string *Ericsson IP-Phone* is used in DHCP option 43 and:
  - if vendor class (option 60) is used, **it is mandatory to initiate the new vendor class for Aastra IP-Phone**.
  - if vendor class (option 60) is **not** used, the string *Ericsson IP-Phone* can be kept in the DHCP server. The telephones can handle both strings (but it is not allowed to have both strings in the same option 43 structure).

The recommendation is to enter vendor classes in the DHCP server, one vendor class for Aastra IP-Phone and another for Ericsson IP-Phone (in case of new installation it is sufficient with only the first one). The vendor specific information tags shall be equal within the two vendor classes. See also section 7.11.3.1 Vendor Class Identifier on page 31.

7.11.4 Microsoft® Windows® 2003

Example of settings in Microsoft® Windows® 2003 server.

7.11.4.1 Define Vendor Class

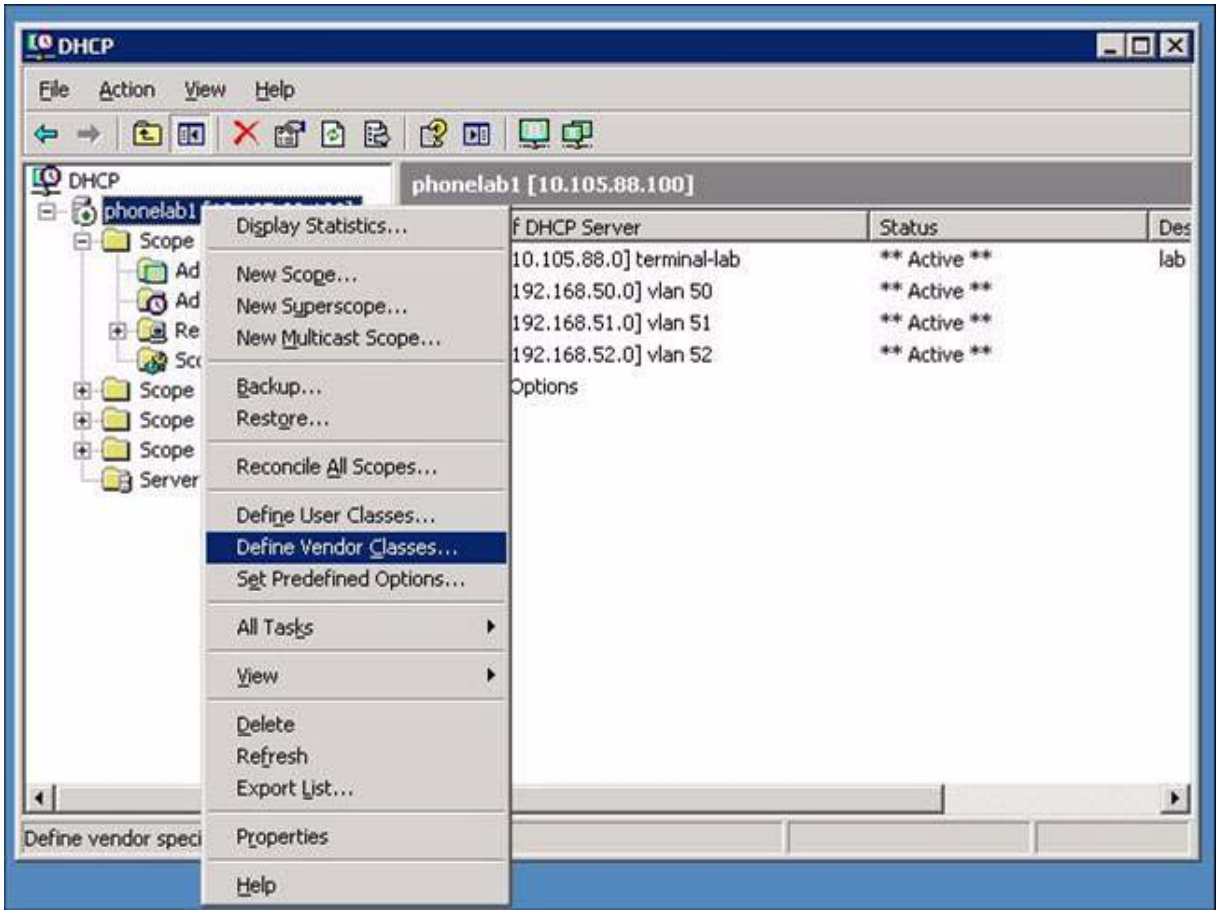


Figure 43:Define Vendor Classes

Select *Define Vendor Classes* to get the menu where the vendor classes are entered.

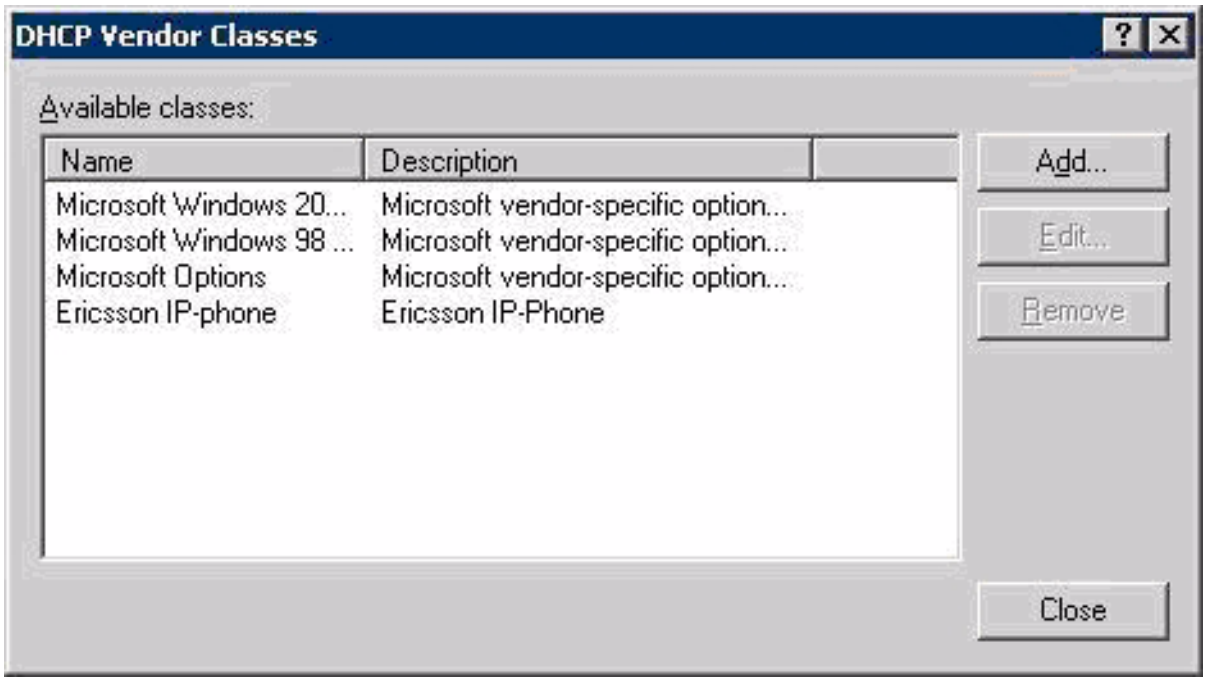


Figure 44: DHCP Vendor Classes

If the vendor class Aastra IP-Phone does not exist, press *Add* to create the new vendor class. In the next menu the ID string *Aastra IP-Phone* has to be entered:

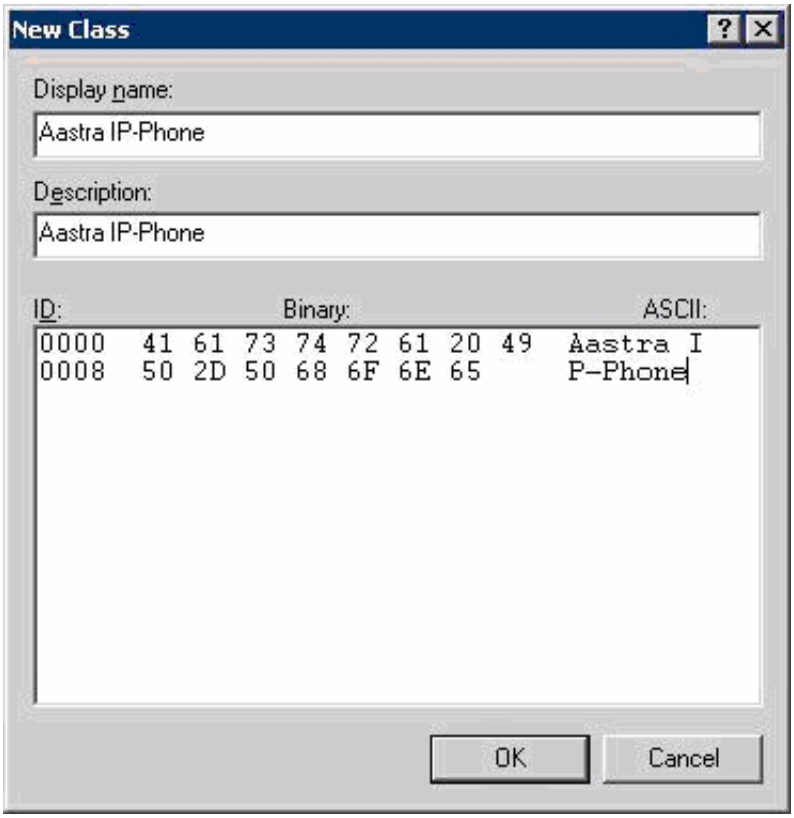


Figure 45:Add Vendor Class

It is possible to move the cursor between the Binary and the ASCII area to make it easier to enter the ID data.

When the data has been entered, press *OK*.

Close the window and proceed to set predefined options.

In some scenarios, the vendor class *Ericsson IP-Phone* has also to be initiated, see section 7.11.3.2 Vendor Specific Information Field on page 32.

The *Standard* vendor class shall be avoided. It is sent out to all devices that ask for option 43 data and if the device does not interpret the data correct, it can cause problem.

7.11.4.2 Set Predefined Options

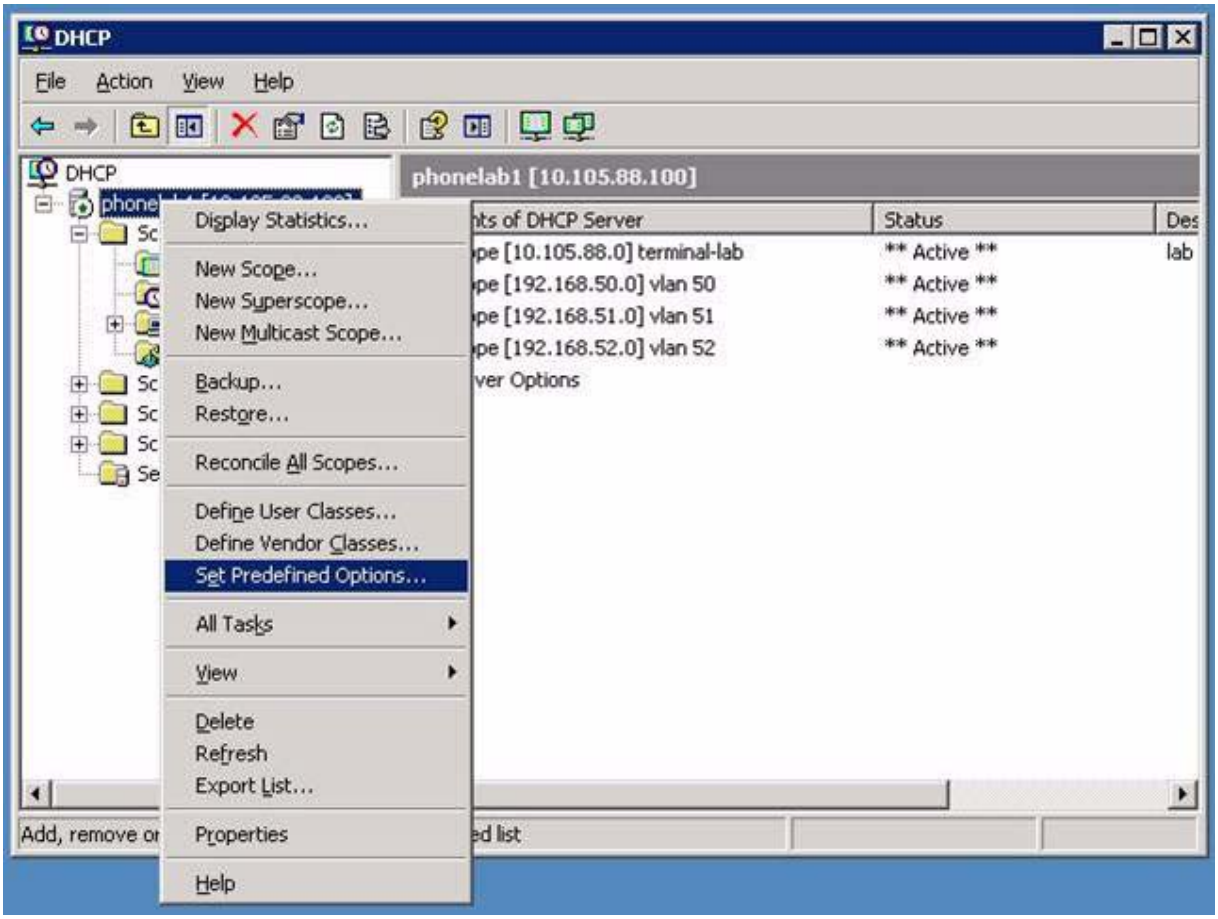


Figure 46:Set Predefined Options

Select *Set Predefined Options* to get the menu to enter option 43 data.

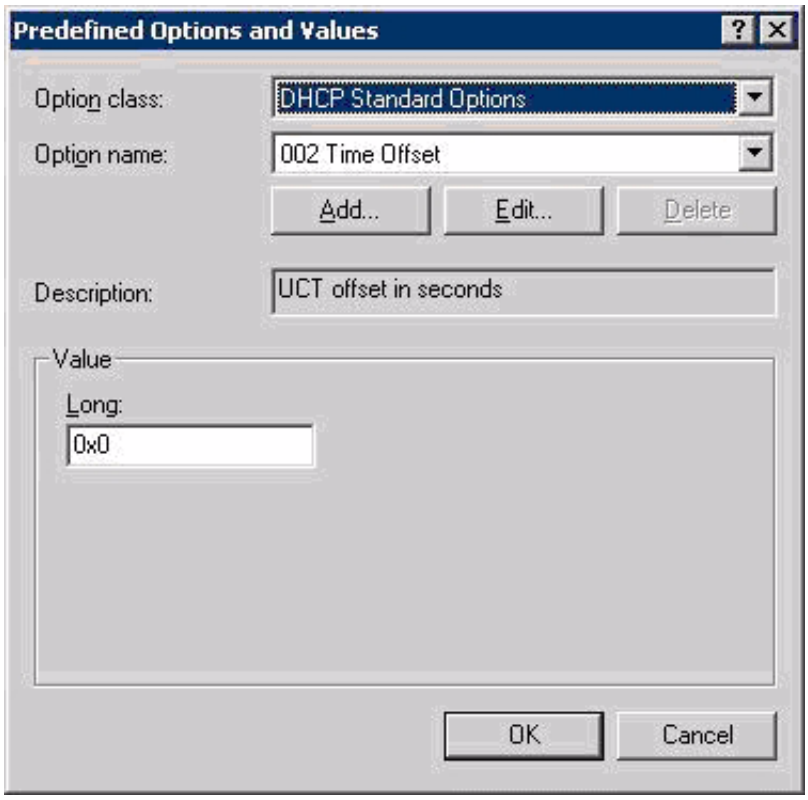


Figure 47: Predefined Options and Values

Select Aastra IP-Phone in the drop down list in the Option class field and press the Add button.

The next menu is shown below:

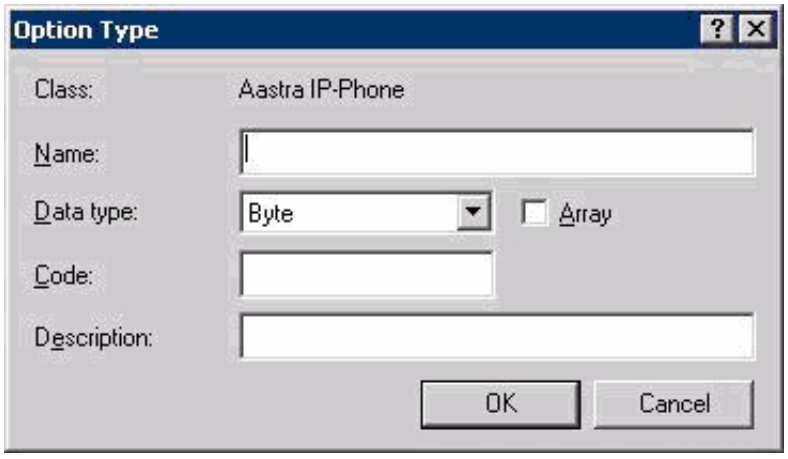


Figure 48: Option Type

This is the default view and data has to be entered manually:

Name: Enter *Vendor specific info*
Data type: Select *Binary* in the drop down list
Code: Enter 43
Description: Can be left empty

The filled in dialog will look like:

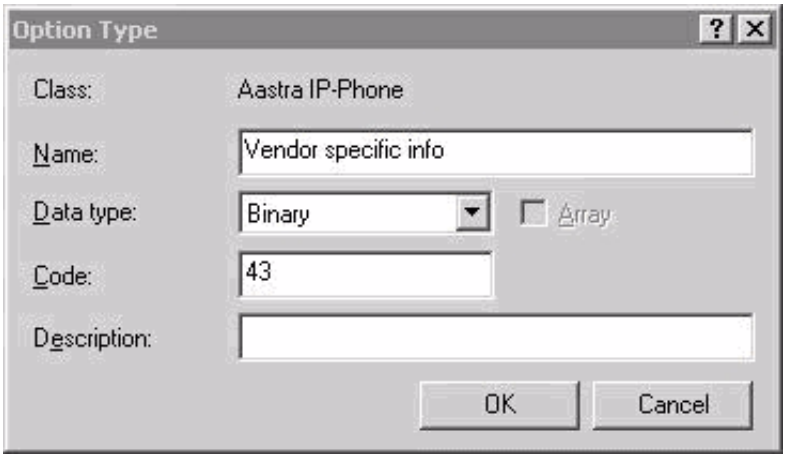


Figure 49: Filled in Option Type Dialog

Press *OK* and the window with Predefined Options and values will occur again. Press *OK* again and the menu will be closed.

7.11.4.3 Set Scope Options

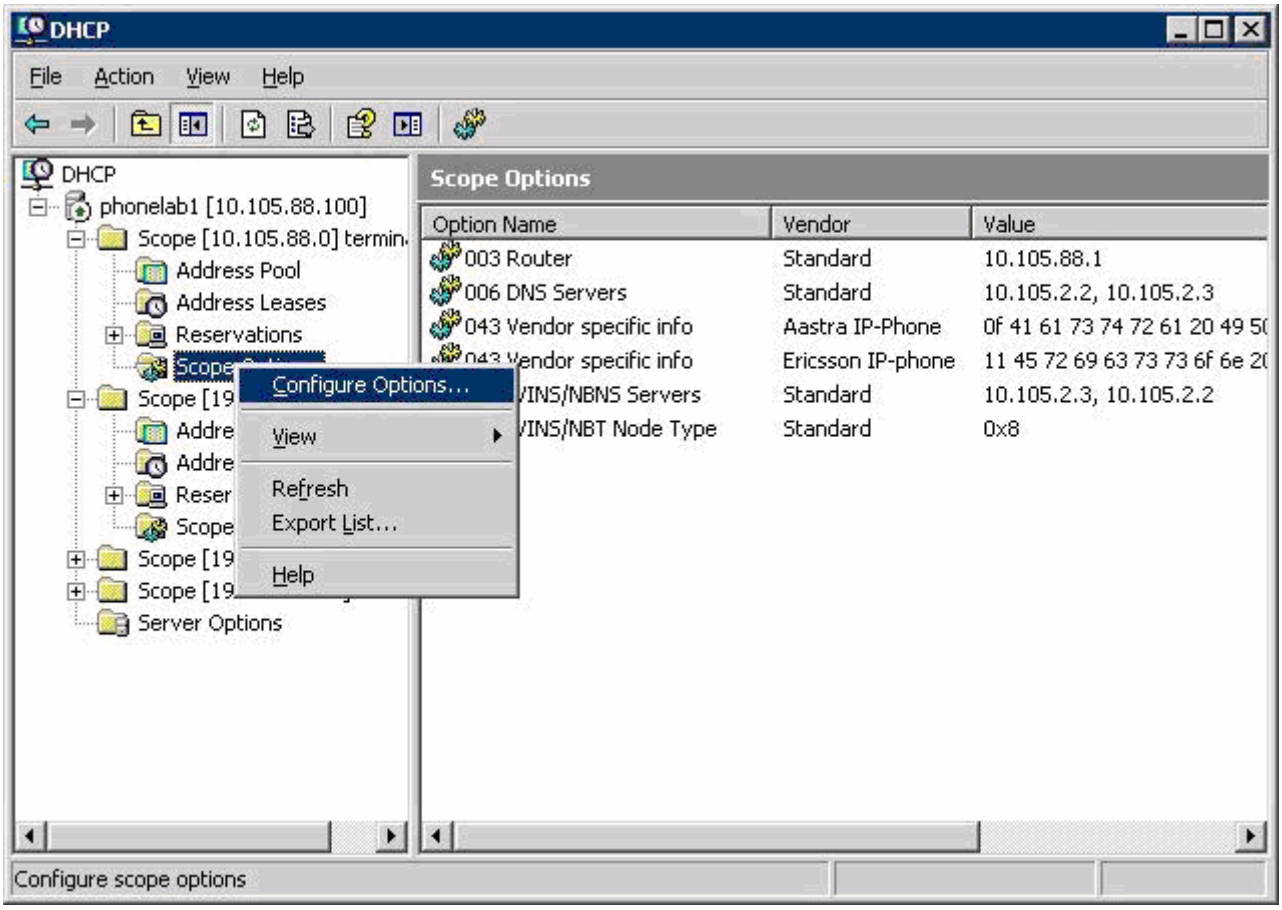


Figure 50: Configure Options

Select *Configure Options*.

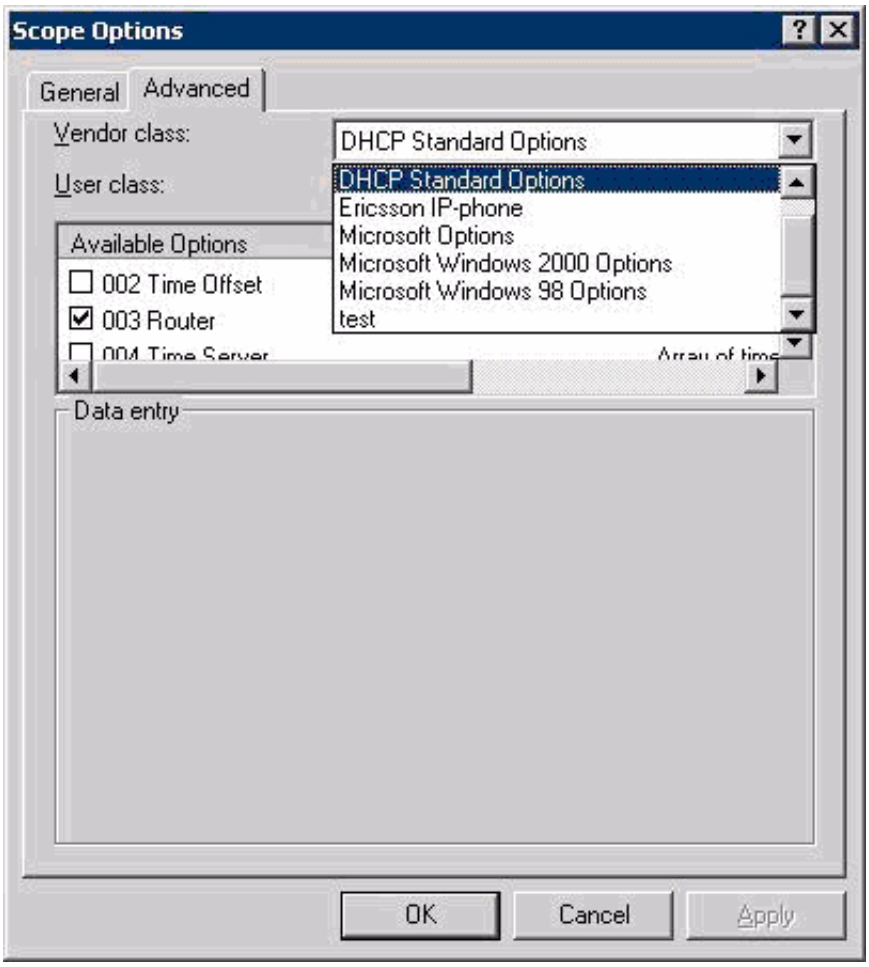


Figure 51:Scope Options

Select *Advanced* tab and scroll in the *Vendor class* field until Aastra IP-Phone is selected. Press *OK*.

Next menu is where the ID strings and the tags are set, according to the figure in section 7.11.3.2 Vendor Specific Information Field on page 32.

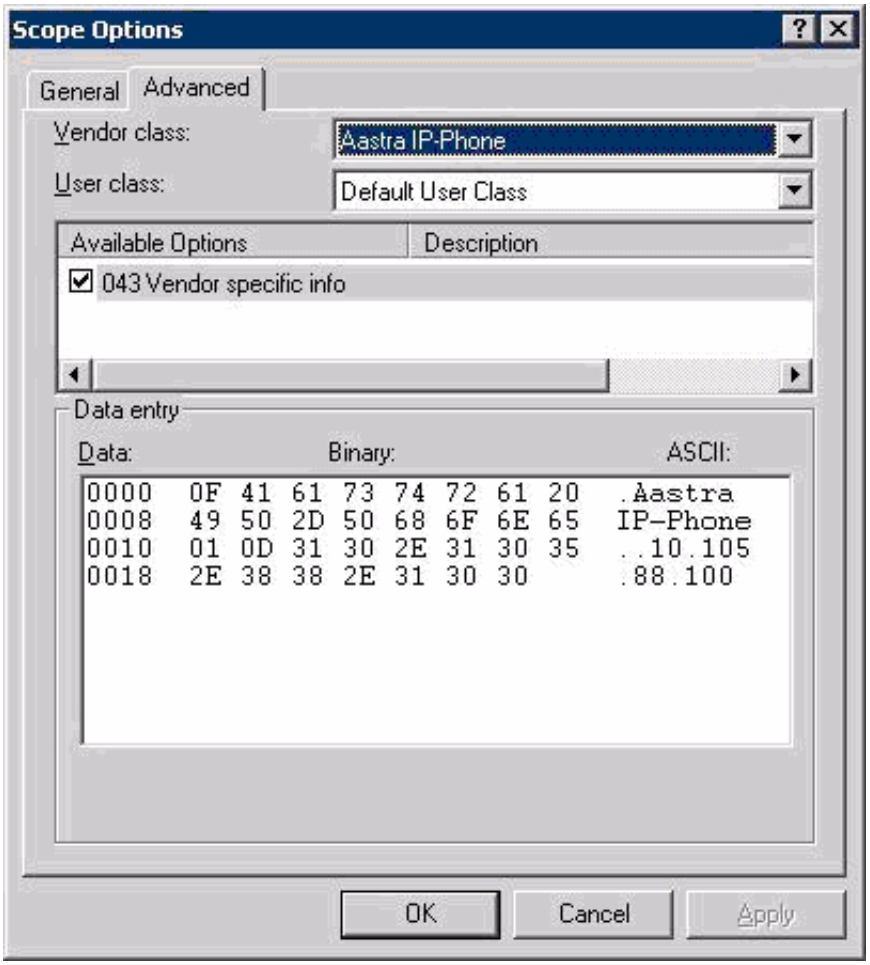


Figure 52:Windows® 2003 server DHCP settings

It is possible to move the cursor between the Binary and the ASCII area to make it easier to enter the option 43 data.

This example shows that the total length of the vendor specific information is 0x1F, the length of the ID string is 0x0F and the string is Aastra IP-Phone, The next byte 01 is the tag for the SW server's IP address, 0x0D is the length and then follows the IP address (10.105.88.100). If more tags than tag 01 for the SW-server is needed, add the additional tags according to the figure in section 7.11.3.2 Vendor Specific Information Field on page 32.

The picture below shows an example how option 43 can look like when two vendor classes are initiated.

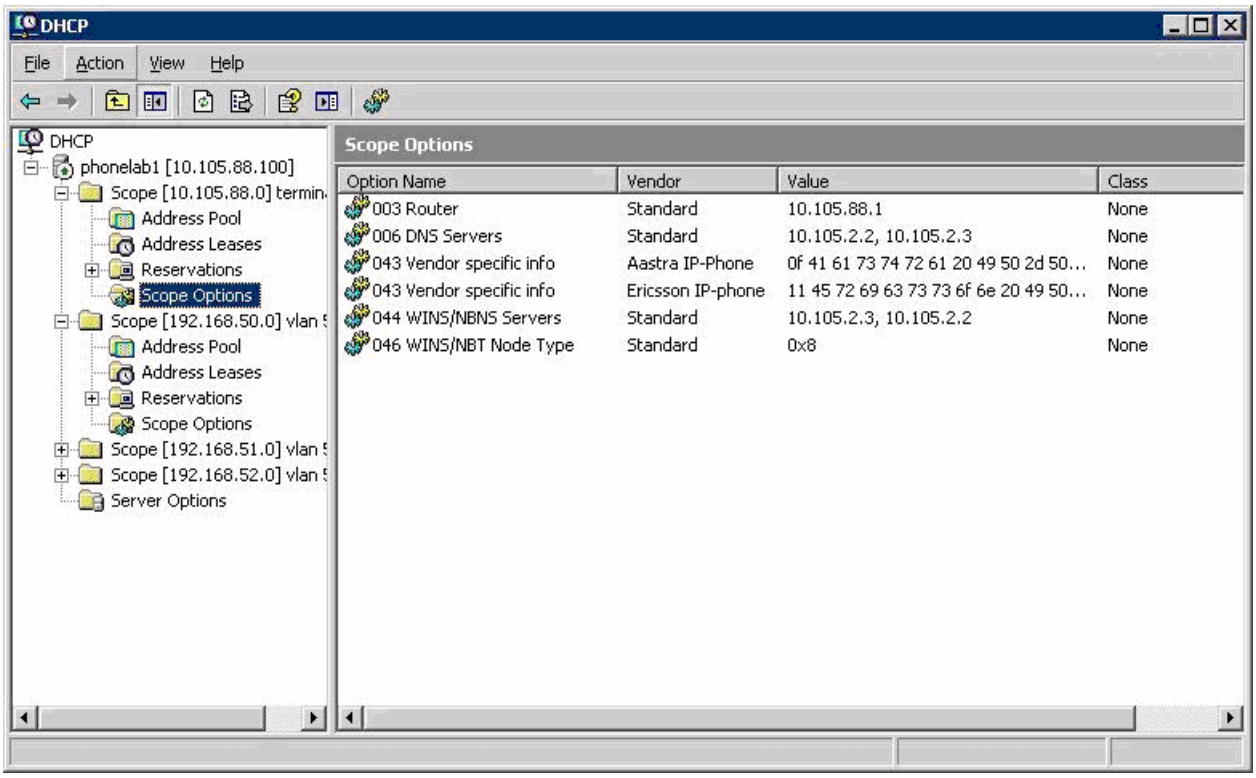


Figure 53:Two Initiated Vendor Classes

7.11.5 Linux DHCP settings

Example of settings in the Linux server:

```
subnet 192.168.6.192 netmask 255.255.255.192 {

option routers 192.168.6.254;

# class "Aastra IP-Phone" {
# match option vendor-class-identifier;
#}
# class "Ericsson IP-Phone" {
# match option vendor-class-identifier;
#}
if substring (option vendor-class-identifier, 0, 15)
= "Aastra IP-Phone"
{
      option vendor-encapsulated-options "\x0fAastra
IP-Phone\x01\x0b192.168.0.1\x04\x16aastradomain.aast
ra.se\x05\x03452";
```

```
} else if substring (option vendor-class-identifier,
0, 17) = "Ericsson IP-Phone" {
       option vendor-encapsulated-options
"\x11Ericsson IP-Phone
\x01\x0b192.168.0.1\x04\x16aastradomain.aastra.se\x0
5\x03452";
}
#
# DHCP settings continued
```

Example when using Vendor Class and the IP address for the sw-server is 192.168.0.1, the telephony domain is *aastradomain.aastra.se* and the VLAN identity is 452.

7.12 Diffserv

Diffserv is a model for handling of priority, based on the type of service (TOS) field in the IP packet heading. For the definition of Diffserv 54 Diffserv octet on page 42

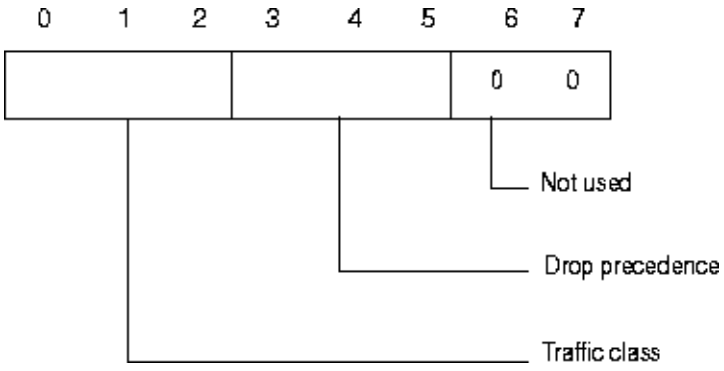


Figure 54:Diffserv octet

The default value for *voice packets* is Expedited Forwarding (EF) which is 101110 (bit 0-5).

The default value for the *signalling packets* is for Traffic class = Class B and Drop precedence = Medium drop precedence (010100 bit 0-5).

It is possible to change the values for Diffserv in the phone via the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

7.13 IP Phone Administrator Tool

The tool *IP Phone Administrator* is used to monitor the DBC 42x 02 IP phones in the network. This is useful in the following cases:

- to find the IP address to the IP phones and especially to the phones without a display.
- to get an overview of all registered and not registered phones
- to see the firmware version in both registered and not registered IP phones

An alternative to *IP Phone Administrator* is to use the SNMP client in the telephone, see section 7.13.2 SNMP agent on page 45.

The *IP Phone Administrator* is used in either of two ways depending on the telephony system:

Table 1

MX-ONE TSE	Use the IP Phone Administrator task, which is part of Manager Telephony System. (No separate installation is needed.)
MX-ONE TSW and other platforms	Use the stand alone application (product number CXC 109 0050), see section 7.13.1 Installation of the IP Phone Administrator server on page 45.

Each phone is sending http messages to the IP Phone Administrator server with data and events. The sent data are e.g. the MAC address, the IP address, the hardware and firmware version and the extension number. The events that can be sent are: the phone has started, the phone is registered or not registered toward the PBX.

The *IP Phone Administrator* tool collects all the http messages from the phones and has a Web GUI to present the data for the system administrator

It is possible to enable / disable the sending of these http messages from the phone with a parameter in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

The phones gets the IP address to the IP Phone Administrator server by DNS SRV resource records or via the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

Below is an example of a printout from IP Phone Administrator:

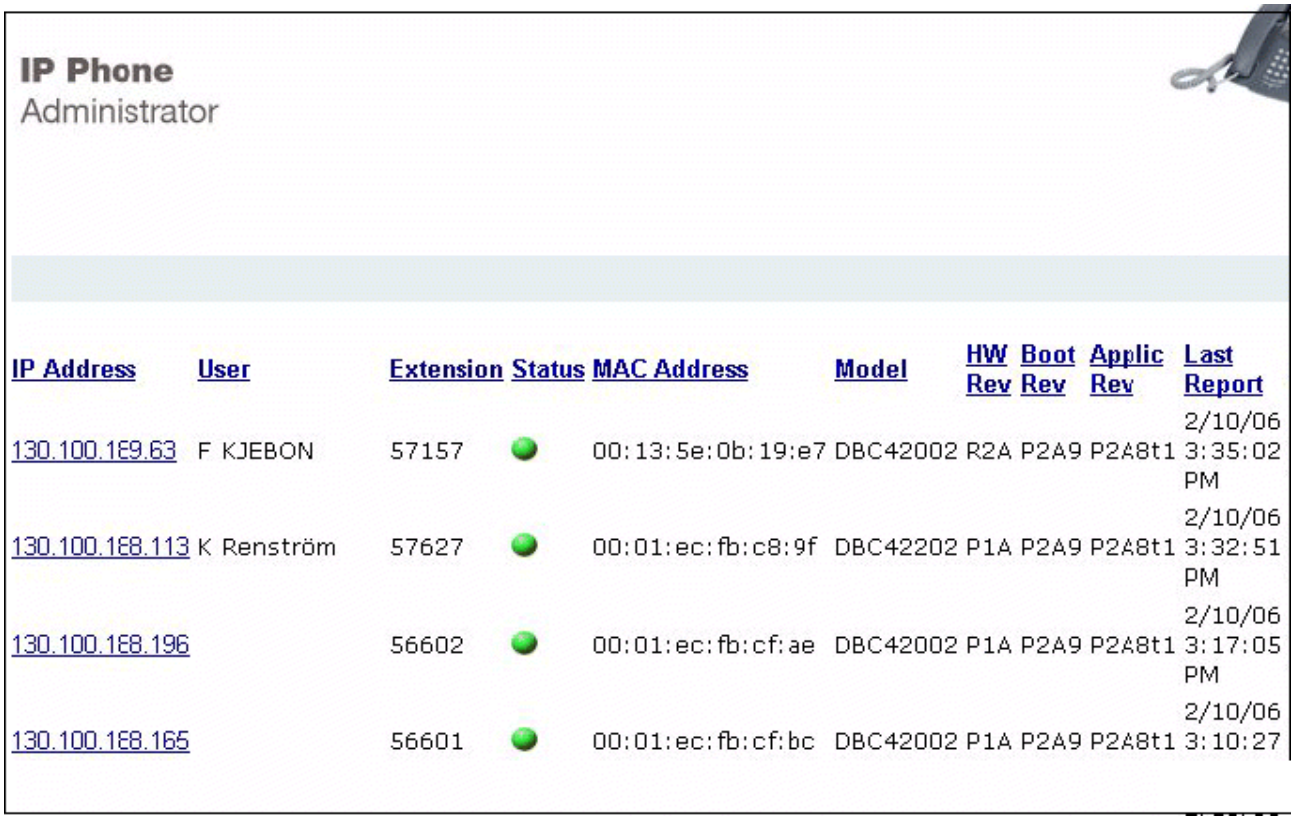


Figure 55:IP Phone Administrator

A log in window will pop up when starting the tool. The user name and the password is set by the system administrator at installation of IP Phone Administrator.

The following columns exist in the GUI:

IP Address
Clicking on the IP address means that the web interface in the phone is opened.

User
The name of the user that is registered or was registered before the phone was logged off. This name is normally received in the phone from the PBX, but can also be the name in the Contacts for the actual extension number.

Extension
Extension number for the user that is registered towards the PBX, or that was registered before the phone was logged off.

Status
An icon in different color is shown:

- Red icon: the phone is not registered towards the PBX.
- Green icon: the phone is registered.

- Grey icon means: no log on attempt towards the PBX has been done
- Yellow icon with an exclamation mark: the phone has tried to register but has got reject back from the gatekeeper.
- Yellow icon: the phone has not reported anything to the IP Phone Administrator since 48 hours.

MAC Address
The MAC address can also be found on the label under the phone.

Model
Type of phone.

HW rev
Hardware revision of the phone

Boot rev
Revision of the bootROM firmware in the phone.

Applic rev
Revision of the application firmware in the phone.

Last report
The time stamp when the phone sent a http message to the IP Phone Administrator. Even if the status in the phone is not changed, the phone sends an update once every 6:th hours.

Uptime
The time since last restart of the phone. The abbreviation **d** means days.

Remove old entries
Removes entries for phones that has not sent any report during the last 48 hours.

7.13.1 Installation of the IP Phone Administrator server

The stand alone tool (product number CXC 109 0050) can be used with other platforms than MX-ONE Telephony Server.

The software can be downloaded from the Service Support Plaza. The files are stored on an Apache Tomcat server. The installation is described in the read me file for the IP Phone Administrator tool.

7.13.2 SNMP agent

There is a built in SNMP (Simple Network Management Protocol) agent in the telephone. When using a port scanning program, the SNMP agent returns the phone model, MAC-address and the hardware and firmware revisions.

The SNMP agent is by default disabled, but can be enabled via the configuration file. In this case it is mandatory to set the *community* string in the configuration file, see description of configuration file for DBC42x.

The MIB (Management Information Base) OID (Object IDentifier) must be 1.3.6.1.2.1.1.1.0.

For a more detailed description of the SNMP agent, see installation instructions for DBC420.

7.14 Gatekeeper address

The IP address of the gatekeeper can be defined either in the configuration file or in the **Settings - Network** menu. To change these settings from the menu, administrator mode must be used. The IP address of the gatekeeper can be set by any of the following methods:

1

Automatic gatekeeper discovery. This is the method to get the IP address automatically, 7.15 Automatic Gatekeeper Discovery on page 47. The gatekeeper and the LAN (enabled for multicast) must support this method. Verify that in the **Network** menu, **Gatekeeper Discovery** is set to Yes or *Default Yes*.

2

In the configuration file. Primary gatekeeper can be defined, see the description for *CONFIGURATION FILE FOR DBC 42X*. Verify in the **Network** menu, that the parameter value is **Gatekeeper Discovery (Auto(No))**.

3

In the configuration file. Secondary gatekeeper can be defined, which will be used when the primary fails, see the description for *CONFIGURATION FILE FOR DBC 42X*. Verify that in the **Network** menu, that the parameter value is **Gatekeeper Discovery (Auto(No))**.

4

Manually entered. Verify that in the **Network** menu, **Gatekeeper Discovery** is set to **No**. Enter the administrator mode. Select the **Gatekeeper** menu to enter the IP address.

5

Backup gatekeeper: the IP address of the backup gatekeeper is defined in the configuration file, 7.37 Backup gatekeeper for branch offices on page 75.

The table below shows which method that will be used depending on the settings in the menus and in the configuration file. The digits refer to the list above.

Table 2

Settings in menus	Backup GK Yes	Backup GK No	Settings in the configuration file
GateKeeper discovery (Yes)	1,5	1	Any value
GateKeeper discovery (No)	4,5	4	Any value
GateKeeper discovery Auto (Yes)	1,5	1	GK discovery = Yes
GateKeeper discovery Auto (No)	2,3,5	2,3	GK discovery = No. Primary and secondary choice available
GateKeeper discovery Auto (No)	4,5	4	GK discovery = No. Primary and secondary choice not available

7.15 Automatic Gatekeeper Discovery

Only certain gatekeepers (PBXes) have support for Automatic gatekeeper discovery, for example MX-ONE TSW.

Automatic gatekeeper discovery is a method to find a gatekeeper (PBX) to register to. When this method is used, the IP phone sends a multi-cast message (Gatekeeper Discovery Request) and waits for a confirmation. Several confirmation messages can be received.

The phone can send the domain name to inform the gatekeeper which domain the phone belongs to. The domain name can be received from DHCP 7.11.2 Data from DHCP on page 30 or from the configuration file. The domain name provided by DHCP has priority over the domain name defined in the configuration file.

The identity of the gatekeeper to which the phone is to be registered can be defined in the configuration file. See data identifier **GatekeeperID** in the description for CONFIGURATION FILE FOR DBC 42X.

The use, or not, of automatic gatekeeper discovery can be defined in the configuration file and in the settings menu. By default the phone uses the value defined in the configuration file.

7.16 HLR Redundancy

HLR redundancy is a function in the MX-ONE TSE system. If the Line Interface Module (LIM), where the data for the extension (Home Location Register) is stored, becomes unreachable, a temporary HLR will be created in another LIM and the IP extension can register towards this LIM.

The functionality of HLR Redundancy is different dependent on which mode the phone is used in.

7.16.1 HLR Redundancy in H.323

The information in this part is valid for H.323.

7.16.1.1 Prerequisites

The HLR redundancy feature will only work when:

- Automatic gatekeeper discovery (with multicast) is **not** used
- The gatekeeper address is **not** set manually
- Backup gatekeeper is **not** used (in branch office scenarios)
- In the configuration file of the phone, both primary and secondary gatekeeper (GK) address must be defined.

7.16.1.2 Change-over to Temporary HLR

The phone will use the primary GK address towards the entry GK in MX-ONE, and receive a list of the GKs to be used. Alternatively the entry GK could accept the registration directly. If a list is received, the phone will try to register according to the list.

When the server (LIM) with the ordinary HLR becomes inaccessible, there are two different cases:

- The phone was registered in the LIM of the ordinary HLR. The terminal will not receive any reply to the keep-alive check and will then try re-registration to the secondary GK, according to the configuration file. A temporary HLR will be created in the LIM where the registration can be accepted.
- The phone was registered in another LIM than in the ordinary HLR LIM. This will happen if load distribution was used when trying to register to the primary GK. A temporary HLR will be created in that other LIM (where the phone was registered).

7.16.1.3 Change-back to Ordinary HLR

If the LIM of the ordinary HLR becomes available again, the periodic keep-alive check request will be rejected by the GK (in the LIM of the temporary HLR). The terminal will request a registration to the primary GK, that is, in the LIM of the ordinary HLR.

The phone will then re-register according to the configuration file, that is, to primary and secondary GK, in that order.

7.17 Domain name

The domain name is used:

- In the function Automatic gatekeeper discovery to find a PBX to register to, 7.15 Automatic Gatekeeper Discovery on page 47.
- When several configuration files will be used, 7.8 Several configuration files on page 22. This domain name cannot be defined as a parameter value in the configuration file.
- In the registration request message when the gatekeeper is MX-ONE TSW or MX-ONE TSE.

7.18 Selection of transport address (port numbers)

The tables below show the port numbers used for signalling and media in the phone. It is the receiving port numbers in the phone that are shown.

Table 3 UDP ports used by the phone

Type of signalling	Minimum	Maximum	Comment
DHCP client	68	68	
SNMP	161	161	
RAS/GRQ	1718	1718	Multicast
RAS	1719	1719	

Type of signalling	Minimum	Maximum	Comment
WAP (Push)	2948	2948	Receive from internal WAP server (PBX)
URQ in secure mode	3727	3727	
OMD	5000	5001	For speech
VoIP Recording	7300	7300	
RTP	16986	17012	
RTCP	16987	17013	RTP port + 1
WAP (Reply)	49152	49152	Receive from internal WAP server (PBX)
WAP (Reply)	49153	49153	Receive from external WAP server

Table 4 TCP ports used by the phone

Type of signalling	Minimum	Maximum	Comment
SSH	22	23	Secure Shell
Web Server Port	80	80	Web server in the phone
H.225 secure port	1300	1300	Incoming call to the phone, default value. Can be 1722 if the phone receives this value in RCF.
H.245	1390	1396	
H.225	1720	1720	Incoming call to the phone

Type of signalling	Minimum	Maximum	Comment
H.225 unsecure port	1722	1722	Incoming call to the phone: is 1722 if the phone receives this value in RCF. The default number is 1300.
RAS over TCP	3727	3727	TLS signaling.
Web Browser Port	8080	8080	When using the WAP browser in the phone to access external web pages

Port number for Operator Media Device (OMD): the port number is set in MX-ONE (OPSAI command). The same port number shall be set in the configuration file for the telephone and in the configuration of the Integrated Attendant Workstation, NOW (if applicable).

7.19 Built-in Ethernet switch

These phones have a built-in Ethernet switch with two available ports. One port is used to connect the LAN and the other can be used by a PC.

The phone has support for the IEEE standards 802.1D (except spanning tree) and for 802.1p&Q.

The frames sent from and to the phone (voice and signalling) are handled with higher priority within the switch compared to the frames sent from and to the PC.

7.20 Virtual LAN (VLAN)

The built in Ethernet switch can handle virtual LAN identities and priorities for the LAN port, for the phone port and for the PC port.

The following possibilities to assign VLAN identities exist:

- From **DHCP** in option 43 (only the phone port, but not the PC port). A list of maximum three VLAN identities can be handled, see figure 41.
- From the **configuration file** (both the phone- and the PC port).

- From a **menu** where it can be manually set (both the phone- and the PC port).

It is possible to change the different VLAN options from the menu in the boot sequence. The following menu is shown (for DBC 425 as an example):

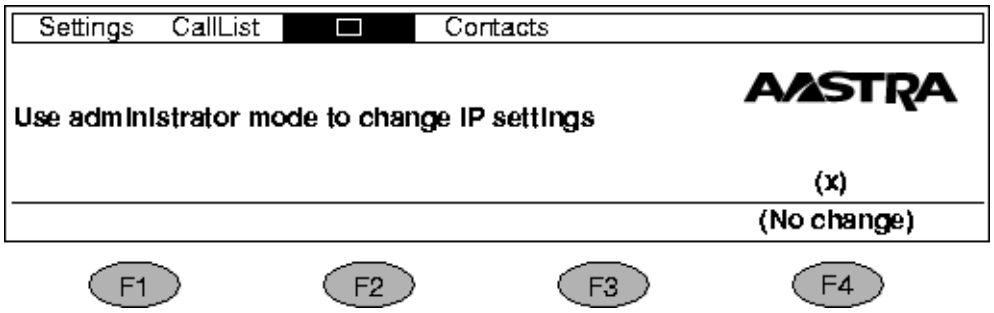


Figure 56: Start up menu in DBC 425

Enter the administrator mode, go down in the **Network** list until the line **VLAN for Phone Port** occurs. Change the value to one of the options below:

- **No VLAN**. VLAN shall not be used, but if a VLAN identity is read from the configuration file, VLAN will be used.
- **Auto.** This is the default value when the phone is delivered from the factory. If the phone receives a VLAN identity list from DHCP (in option 43) or if there is a VLAN identity defined in the configuration file it will be used. The VLAN identity received from DHCP has priority over the configuration file. For more information, 7.20.1 Automatic VLAN detection with DHCP on page 53 and 7.20.2 Assigning the VLAN identity via the configuration file on page 53.
- **Manually**. The manually entered VLAN identity will be used, 7.20.3 Manual setting of the VLAN identity on page 54. If the manually entered VLAN identity shall be used, the [L2QOS] header in the configuration file has to be omitted.

Even when VLAN is not used this parameter can have the default value **Auto**.

Concerning the priority of the frames: For outgoing frames the following priorities will be set at level 2 for each frame by default, when VLAN is used:

- For frames **originating in the phone** the default value will be 6, meaning voice traffic with less than 10 ms latency.
- For frames **originating in the PC** the default value is 0, meaning best effort.

The priorities can be changed via the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

7.20.1 Automatic VLAN detection with DHCP

Prerequisites on the LAN

When the phone is connected to a layer 2 switch, the switch will add the IEEEE 802.1Q header, to untagged frames with the default VLAN identity and forward the frames. The first layer 3 switch must be initiated for DHCP relay and having an ingress port with an IP address on each of the offered VLANs. The address to the DHCP server must be set in the layer 3 switch.

When the layer 3 switch has received a *DHCP discover* message, it will forward this packet to the DHCP server adding the IP address of its ingress port corresponding to the VLAN. It is this address information that informs the DHCP server to which IP subnet that this phone is to be assigned to.

Description of when a VLAN identity list is received from the native LAN

At installation (and hardware reboot) the phone asks for a temporary IP address from DHCP by initiating the DHCP negotiation with untagged messages (native LAN). The relay agent adds the address of its ingress port corresponding to the native LAN. DHCP provides the temporary IP address together with the VLAN identity list. The phone releases the temporary IP address.

Then the phone uses the first VLAN identity in the list and sends a new tagged request to the DHCP server. The relay agent adds the address of the ingress port corresponding to the **selected** VLAN. If there is any available IP address, the DHCP server provides this address to the phone. If there is no available IP address for this VLAN, the phone takes the next VLAN id in the list and asks for an IP address.

If there is no IP address available in any VLAN in the list, the phone will ask for an IP address in the native LAN.

Reboot

It is possible to specify in the configuration file whether the telephone should retain the previously used VLAN identity after reboot, or whether it should start a new automatic VLAN detection procedure. See description for Configuration file for DBC42x.

To change the VLAN identity:

See description for Configuration file for DBC42x.

7.20.2 Assigning the VLAN identity via the configuration file

The description of the parameters, see the description for *CONFIGURATION FILE FOR DBC 42X*.

The configuration file is read from the native LAN

At installation (and hardware reboot) the phone asks for an IP address from DHCP by initiating the DHCP negotiation with untagged messages (native LAN). DHCP provides the IP address but no VLAN identity list. The phone reads the configuration file, but in this case when no VLAN identity list is received from DHCP, a software reboot is done automatically in the phone to get the IP address valid for the tagged VLAN defined in the configuration file.

The configuration file is read from the VLAN

At installation (and hardware reboot) and the configuration file is available in the VLAN but not in the native LAN, the VLAN identity must be set manually in the boot menu.

7.20.3 Manual setting of the VLAN identity

Set the VLAN identity from the menu in the boot sequence. To use the manual entered VLAN identity all the time, disable the VLAN settings in the configuration file.

7.21 Security

There are two security features:

- LAN access control, see 7.22 LAN access control (according to IEEE802.1x) on page 58.
- VoIP signalling with TLS and media encryption with SRTP. This is described in this section.

The TLS and SRTP support can be enabled/disabled from the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X* section SECURITY.

In addition there is a security policy in the telephony system which also affects the behavior of the IP phones. For MX-ONE Telephony Server, see the description for *SECURITY*.

The security policy is checked at the registration time. Once the phone is registered, all kinds of calls can be established from a security perspective.

When a secure IP to IP call is established, with TLS and SRTP, a secure icon (a padlock) is shown in the display. For all gateway calls the secure icon is not shown because the other party can have an un-secure connection. When there is a secure IP to IP call and IP voice recording is active the secure icon is not shown.

7.21.1 Protection of VoIP signalling

The signalling between the DBC 42x 02 IP phones and the gatekeeper is protected by means of TLS (Transport Layer Security) according to RFC 2246.

The TLS protection affects the registration and the call handling. Multicast traffic (automatic gatekeeper discovery) is not protected.

The TLS server (gatekeeper) makes use of a digital certificate to authenticate itself towards the terminal. The terminal authenticate themselves by means of the password (ordinary password to register towards the gatekeeper) sent in the RAS/RRQ message.

TCP port 3727 is used for RAS over TCP.

TCP port 1300 is used for Secure Call Setup. For more information 7.18 Selection of transport address (port numbers) on page 49.

The cipher suite TLS_RSA_WITH_AES_128_CBC_SHA defined in RFC 3268 is used.

TLS is not supported on top of UDP. In order to support TLS protection of the RAS messages these are sent over a TCP connection, opened by the IP phone, after a TLS connection has been set up.

The TLS support can be enabled/disabled from the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

7.21.1.1 Certificates

The digital certificates are in X.509 version 3 format with the file extension **.pem**. For more detailed information about creating the certificate, see *operational directions for Certificate Management* in the CPI library.

In order for the phone to be able to authenticate the server, the phone has a certificate repository with a number of root certificates or trusted certificates (see the table below). These are included in the IP phone firmware in the factory.

It is also possible to add another root certificates beside these by reading in the file with the certificate from the software server. The file must be stored under the folder ***/certificates/H323***, see section 7.9.3 Directory structure on page 25. The path to the certificate file is specified in the configuration file.

Table 5 X.509 root certificates to support TLS server authentication

Certificate Authority	Comment
Baltimore	
Entrust	md5WithRSAEncryption

Certificate Authority	Comment
Entrust	sha1WithRSAEncryption
Equifax CA-1	md5WithRSAEncryption
Equifax CA-2	sha1WithRSAEncryption
Equifax	sha1WithRSAEncryption
Equifax Secure Global eBusiness CA-1	
GTE Cyber Trust	
QuoVadis Root CA2	
SecureSign Root CA1	
SecureSign Root CA2	
SecureSign Root CA3	
Tawnte Premium Server CA	
Tawnte Server CA	
ValiCert Class 1	
ValiCert Class 2	
ValiCert Class 3	
VeriSign Class 3	
VeriSign Class 3 - G2	
VeriSign Class 4 - G2	
VeriSign Class 3 - G3	
VeriSign Class 4 - G3	
VeriSign Test Root CA	md2WithRSAEncryption
VeriSign Test Root CA	sha1WithRSAEncryption

7.21.1.2 Registration towards the gatekeeper

At log on the phone promts the user to enter the extension number and the password or PIN. If the user do not have a password or PIN, the phone tries to log on to the insecure UDP port1719.

In case the IP phone tries to log on securely but the establishment of the TCP connection fails, this is interpreted as the gatekeeper does not support secure mode. The phone shall back off to RAS over UDP. The possibility to back off to UDP is managed via a parameter in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

During the TLS negotiation, the server will authenticate itself by using a digital certificate, see 7.21.1.1 Certificates on page 55.

In the configuration file there is an option whether the client shall validate the server certificate or not. If the option is enabled but the server does not have a certificate that is signed by one of the Certificate Authorities supported in the phone or if the certificate has expired, it will result in a failed authentication.

There are two options in the configuration file for the password:

- Do not store the password in the phone: The user needs to re-enter the password each time the phone registers towards the gatekeeper, that is, after power failure, network failure, update of firmware. This option is not available for the DBC 420 phone, which do not have the log on option from the key pad.
- Store the password in the phone in the same way as when not using TLS, that is the user only needs to re-enter the password after the phone is manually logged off or when the phone has been logged off after the extension number is used by another IP terminal.

7.21.1.3 Call Setup and Call Control

When the IP phone that is registered securely, sets up a call using H.225 Q.931 messages, it sends the requests to TCP port 1300 instead of TCP 1722.

In order to negotiate the capability of the call, an H.245 negotiation takes place on a new TCP connection between the terminal and the gatekeeper. The TCP port to be used is negotiated during the H.225 signaling. The TCP connection can be initiated by either part. This TCP connection is protected by means of TLS as well.

This implies that during a call there can be three TCP connections existing between the terminal and the gatekeeper.

7.21.2 UDP Filtering

All the UDP ports that are not used, can be blocked for security reasons. For a description of all UDP and TCP ports, see 7.18 Selection of transport address (port numbers) on page 49.

The default value is that the UDP filtering is enabled, but can be disabled with a parameter in the configuration file, see description of Configuration File for DBC 42x.

7.21.3 TCP filtering

All the TCP ports that are not used, can be blocked for security reasons. For a description of all UDP and TCP ports, see 7.18 Selection of transport address (port numbers) on page 49.

The default value is that the TCP filtering is enabled, but can be disabled with a parameter in the configuration file, see description of Configuration File for DBC 42x.

7.21.4 SRTP

Secure RTP, SRTP (RFC 3711), is supported by DBC 42x 02 phones. The supported encryption algorithm is AES 128 (Advanced Encryption Standard) in counter mode for SRTP and SRTCP. HMAC_SHA1_80 is supported for SRTCP.

Media encryption is negotiated using H.245 i. e. both the capability as well as the keys. (The key negotiation phase is based on H.235.8).

The following codecs have SRTP support: G.711 A-law, G.711 µ-law, G.723.1, G.729a and G.729ab.

Beside the possibility to enable/disable TLS and SRTP via the phone configuration file, SRTP can be temporary disabled for a certain phone via a SSH command.

7.22 LAN access control (according to IEEE802.1x)

The IEEE802.1x standard is used for port access control authentication. The LAN must support IEEE802.1x signalling and there must be a RADIUS server handling the authentication, according to EAP-MD5. The system administrator, or the end-user, enters the user identity and the password into the phone and if the authentication is successful, the phone gets access to the LAN and continues with the ordinary boot sequence.

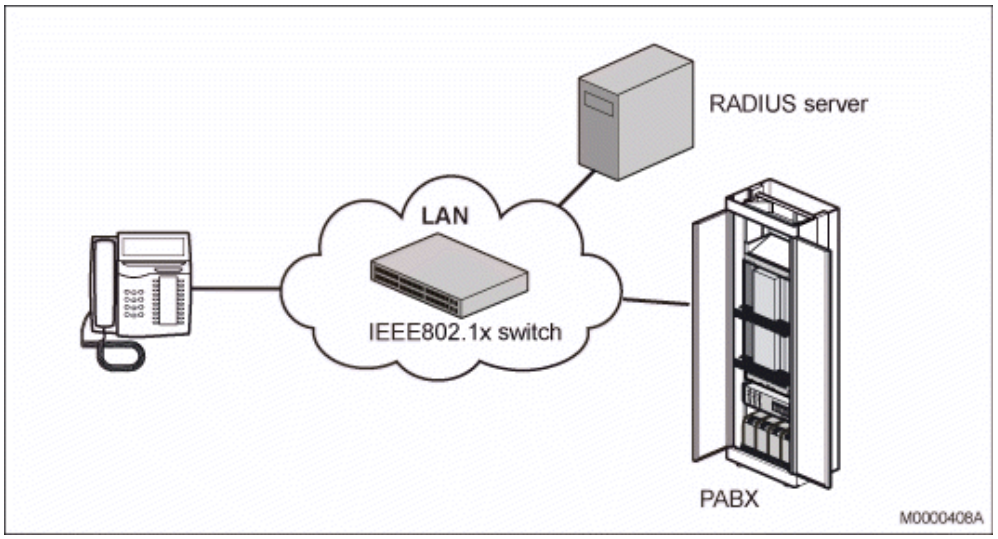


Figure 57:Components in LAN access control

Before the authentication the phone cannot get access to the LAN or even get the IP address from the DHCP server. The authentication has to be performed periodically within 5 minutes (maximum).

If the LAN does not support IEEE802.1x, the phone will start in the ordinary way.

If a PC shall be connected to the PC port in the phone, the phone supports that the PC and the phone are authenticated independent of each other.

Note: The LAN switch must support that two devices are authenticated independent of each other on the same LAN port.

7.22.1 Configuration of LAN access control

The default value is that the phone shall automatically detect if IEEE802.1x is enabled in the LAN. In the boot menu, it is possible to change the following values:

- **Auto**. The phone will initiate IEEE802.1x signalling in the boot sequence and if IEEE802.1x is enabled in the LAN, the phone enables this function.
- **No**. The phone disables the IEEE802.1x function. If IEEE802.1x is enabled in the LAN and the parameter value is set to **No**, the phone cannot access DHCP and the configuration file cannot be read.

The parameter values set in the boot menu is valid until the configuration file is read by the phone. If the values in the boot menu shall be valid even after the configuration file is read, the corresponding parameters in the configuration file shall be disabled.

In the configuration file the following parameters exist:

- LAN access control (Auto / No)
- LAN access control user identity and password shall be stored in the phone or not.
- LAN access control user identity and password shall be valid for *the phone* or for *the end user*. In the first case the phone shall not log off from the LAN when register with a different extension number towards the PBX. In the second case, the phone shall log off from the LAN when entering a different extension number.
- LAN access control user identity and password, which can be used when all the phones shall have the same user identity and password.

For more details, see the description for *CONFIGURATION FILE FOR DBC 42X* section 802.1X.

7.22.2 Examples of configuration

One typical case with basic level of security can be that all phones have the same user identity and password. The configuration of all the phones must be done via a switch where IEEE802.1x is disabled. Using the default values the following configuration will be needed:

- Automatic IEEE802.1x detection (default).
- Define the user identity and the password in the configuration file.
- The user identity and password are stored in the phone (default).
- The user identity is valid for the phone, which means that the end-user can change extension number towards the PBX, without having to enter a new LAN access user identity and password (default value).
- Start the phone via the IEEE802.1x disabled switch. The phone will read and store the user identity and password from the configuration file.
- Set out the phone to the end-user and start it via the IEEE802.1x enabled switch.

Another typical case with a higher level of security is that each phone has individual user identity and password:

- Automatic IEEE802.1 detection (default).
- The user identity and password are stored in the phone (default).
- The system administrator or end-user have to enter the LAN access control user identity and password when starting the phone only the first time.

- The user identity is valid for the phone, which means that the end-user can change extension number towards the PBX, without having to enter a new LAN access user identity and password (default value).

7.22.3 Authentication to the LAN

The authentication process starts in the boot sequence with that the phone prompts the user to enter the user identity and password, 7.1 How to start a new phone on page 8.

At restart of the phone, and when the user identity and password are stored in the phone, the ordinary restart procedure is done which means that the user does not have to do anything.

7.23 Access the phone from a PC

For maintenance of the terminal, the system administrator can from a PC access the phone in one of the following ways:

- Web interface. This interface is recommended.
- SSH (Secure Shell). This interface is similar to a Telnet interface, but the connection is secure.

  In the maintenance PC, a SSH client must be used. There are a number of free-ware clients, the must popular is PuTTY for PCs with Windows®.

  The default encryption keys are used and not possible to change.

For a description of these interfaces, see maintenance instructions for *IP TELEPHONE DBC 42X*. See also 7.24.1 Password for maintenance on page 62.

The *end-user* can also access the phone via the web interface. This interface is described in the directions for use for each platform. See also 7.24.2 Web interface password for the end user on page 62.

7.24 Passwords

There are different passwords used in the phone:

- to register the phone to the gatekeeper, see 7.2.1 Starting a phone in a LAN with a DHCP server on page 8 and 7.3.1 Starting a DBC 425 phone in a LAN with a DHCP server on page 14. The recom-

mendation is to use such a password or PIN to avoid that one end-user can log in with another end-user's directory number.

- for maintenance via SSH or the web interface, to be used by the network administrator or other maintenance personnel, 7.24.1 Password for maintenance on page 62
- for the end user when handling of data in the phone, via the web interface, see 7.24.2 Web interface password for the end user on page 62.
- LAN access control authentication, see 7.22 LAN access control (according to IEEE802.1x) on page 58.

7.24.1 Password for maintenance

A network administrator can log on from a PC to an IP phone via SSH. It is also possible to access the phone from a web browser via the web interface. It is the same password for both SSH and web browser access. The default password is **Telephone**. This password can be changed by using the following procedure:

- Log in to one IP phone via SSH.

  VxWorks login: admin

  Password: Telephone

  For more information about the functions when using Telnet, see the maintenance instructions for *IP TELEPHONE DBC 42X*.
- Enter the command **encryptPasswd "new wanted password"** (quotation mark must be used). The password must be at least eight (and maximum 40) characters. Letters A - Z, a - z and digits can be used.
- Write the generated encrypted password with the data identifier **AdminPassword** in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*. Store the updated configuration file on the SW server
- Next time the IP phones read the updated configuration file the new password is valid.

7.24.2 Web interface password for the end user

The end user can use the web-browser in a PC to access the web interface in the IP phone. The purpose is to set data in the phone like entries in the Phone Contacts, the call list etc.

This password is the same as the password or PIN to register the phone to the PBX. If there is no password or PIN initiated in the PBX, it is

possible to use the default password is **Welcome** to log in to the web interface. But in this case a parameter must be changed in the configuration file to enable the default password option.

7.25 Software version

It is possible to check the software versions in the phone by pressing the keys **C** (clear key), ***** and **4** simultaneously for at least one second.

The program revisions are shown in the display. To check that all pixels are working, press the **C**-key. To get to the original mode again press the # key.

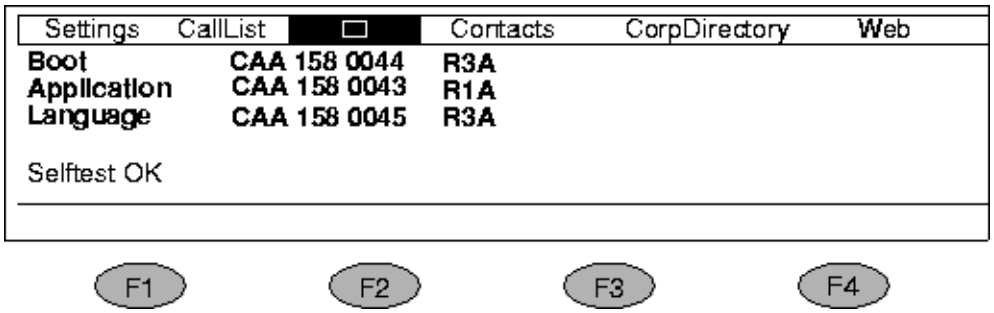


Figure 58: Display of software versions in DBC 425

In the example above the version of the boot is R3A, the application R1A and the language file R3A.

If there is a fault in the configuration file an error message is shown in the display, see maintenance instructions for *IP TELEPHONE DBC 42X*.

7.26 Keys for DBC 422

7.26.1 Function-keys

In the Settings menus, the function-keys have the following functions:

Clear key

/C

Cancel. Leave the current menu without changing anything.

Exit. Leave the current menu and go up one level in the menu hierarchy.

Speaker key

/

Select the current menu.

Save the entered data.

Volume keys

+ key is used to go down one row in a listancel.

- key is used to go up one row in a list and also to erase the latest entered character.

The figure below shows an overview of what the keys are used to.

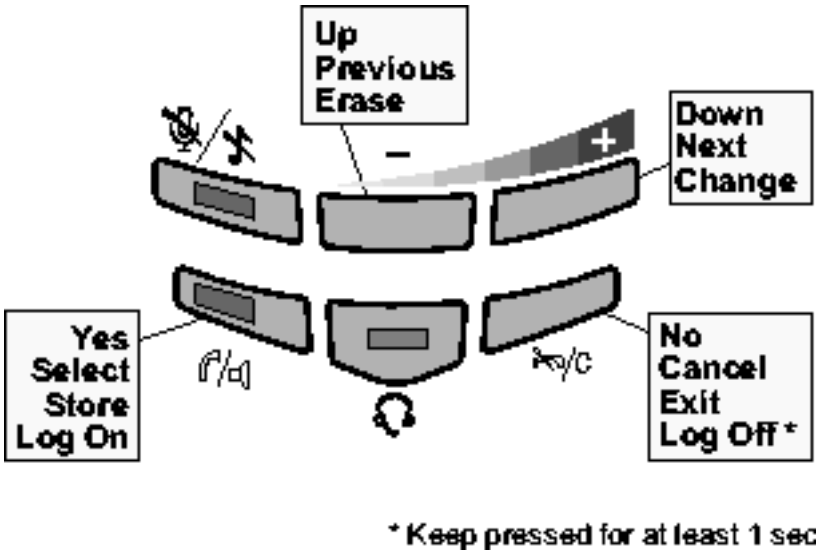


Figure 59:Hidden Key Functions

The functions assigned to the programmable function keys are possible to move and remove, see 7.34.3 Allocation of function keys on page 70.

7.27 Keys for DBC 425

The functions assigned to the programmable function keys are possible to move and remove, 7.34.3 Allocation of function keys on page 70.

7.27.1 Soft-keys

In the Settings menus, the soft-keys have the following functions:

Cancel (F1)

Leave the current menu without changing anything. Cancel is used in menus where it is possible to change data.

Exit (F1)
Leave the current menu and go up one level in the menu hierarchy. Exit is used in menus where it is not possible to change data.

Erase (F3)
Erase the latest entered character.

Select (F4)
Select the current menu.

Save (F4)
Save the entered data.

Down (F3)
Scroll down or to the right in the menu.

Up (F2)
Scroll up or to the left in the menu.

7.27.2 Navigation keys

The navigation keys are used to navigate in the menu bar on the top display row:

<
one position to the left.

>
one position to the right.

Home
the home key to get to the idle menu from all other menus.

7.28 Restart and reboot of the phone

If it is necessary to restart or reboot the phone manually, press the keys **C** (clear key), **mute** and # simultaneously for one or two seconds.

It is also possible to restart the phone and to reset all network settings from the administrator web interface, see maintenance instructions for *IP TELEPHONE DBC 42X*.

After a power failure or a reboot, the phone will log on automatically with the stored directory number.

7.29 Factory default

To set the phone to factory default, the following procedure must be used:

- Enter administrator mode, 7.30 Administrator mode on page 66.
- Press the keys C,*,9 for a second.

The following data will be set:

- Use DHCP
- Auto detection of protocol, which means that the telephone will use the protocol H.323 that is defined in the configuration file
- The phone shall retrieve the IP address to the software server automatically (from DHCP or DNS SRV resource records)
- Automatic VLAN detection
- Automatic detection of LAN access control according to IEEE802.1x
- User identity and password for LAN access control according to IEEE802.1x
- The password is erased
- The contact list is erased
- All the numbers associated to function keys are erased
- All the function keys are placed on the default positions
- Tone ringer character is set to the default value
- Increased hearing level is set to standard

7.30 Administrator mode

The administrator mode is used when IP settings is to be changed. Normally it is the maintenance personnel that handles the IP settings and not the end user.

To enter the administrator mode, use the navigation key to get the settings menu and then press the keys **C** (clear key), ***** and **5** simultaneously for one or two seconds. One ringing signal is heard to indicate that this mode is entered. The administrator mode is valid until exiting from the **Settings** menu.

7.31 Quality of Service (QoS)

It is possible to view the quality of service statistics of the connection for the last 10 calls via the web interface. The statistics shows for example the delay, jitter and number of lost packets. See maintenance instructions for *IP TELEPHONE DBC 42X*.

7.32 Language handling

The language file will be stored on the SW server and is fetched from the server to the phone at start up of the phone. The end user may choose a language other than English using the language menu in **Settings** mode.

It is possible for the system administrator to change text strings e.g. for Absence reasons. See the description for *LANGUAGE FILE FOR DBC 42X 02*.

7.33 Setting Time and Date

Time and date can be set using the following alternatives:

- WAP messages. For PBX's supporting WAP, the time in the IP phone is updated automatically as soon as the phone is registered towards the PBX.
- SNTP (Simple Network Time Protocol). The time in the phone can be set via SNTP, as described below. This method must also be used if security is enabled (with validation of the server certificate) in the phones and in the system.

If the SNTP server is enabled and defined in the configuration file, the phone will take the time from the SNTP server.

7.33.1 Simple Network Time Protocol

When SNTP is available in the LAN, the time and date in the phone are updated automatically when the phone is started and is verified periodically. If SNTP is used, the IP address and the Time zone will be set in the configuration file, see *Configuration File for DBC 44X and DBC 43X*.

If the LAN does not have SNTP or NTP available, a server with SNTP software has to be initiated.

7.34 To change the configuration of the phone

7.34.1 Methods to change the configuration

It is possible to change the configuration in the phone by:

- The menus, see directions for use for each platform.
- The web interface, see directions for use for each platform.
- The configuration file. The configuration file is read by the phone when it is rebooted, either by an order from the PBX or manually or via the web interface, see 7.28 Restart and reboot of the phone on page 65.
  The phones can fetch the configuration file from the SW-server every 24:the hour and this option is enabled via a parameter in the configuration file.

7.34.2 Log on / Log off options

The following options exist:

- **Log off allowed.** The end-user is allowed to log on and log off the phone. This is the default and most common option. It is possible to force all the telephones to log off at a certain time each 24 hour period, but only when the log off option is defined via the configuration file, see 7.34.2.1 Set the log off restriction option in the configuration file on page 69.
- **Default number used**. The phone is always logged on with a default number. The **log off** soft-key is not shown and the end-user cannot log on or off. This option can be used for phones in conference rooms, receptions etc. The default number must have an associated password to avoid logging off by mistake from an other terminal. If the terminal is logged on when option 2 is set, the current number will be the default number. If the phone is logged off, after exiting the **Settings** menu, the system administrator will be prompted to enter the number to be used as the default number.
- **Permit individual log on.** The phone is always logged on, with a default number, as in option above but the end-user can log on with his/her individual number and get the personal categories. This option can be used in a free seating environment. When the end-user logs off the individual number, the phone registers automatically with the default number. The same if the end-user logs on with the individual number from an other terminal. If the end-user forgets to log off, the phone will log off the individual number and will log on with the default number during the night.

The log on procedures are described in the directions for use for respective system.

To change between the different log on / log off options:

- Select the **Settings** menu.
- Enter the administrator mode.
- Select the **Log Off Restrictions** menu.

For DBC 425, the following menu is shown:

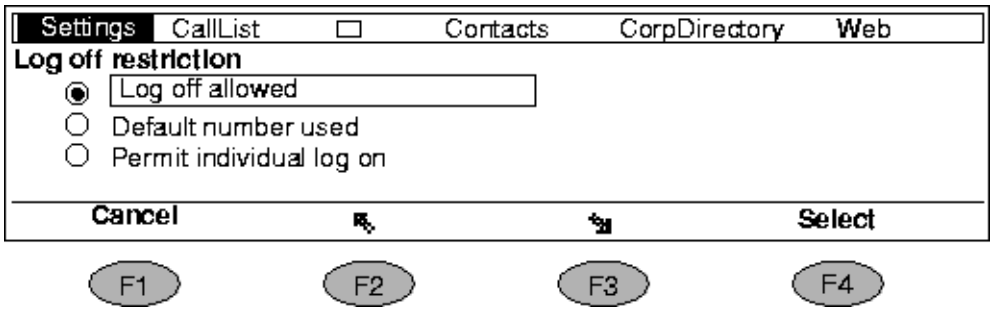


Figure 60:

Select the wanted option.

It is also possible to use the web interface to change the log on options.

If the default number is to be *changed,* there are two possibilities:

- Enter the administrator mode in **idle** mode and the log off menu will appear. Press **Log off** and log on with the new default number and the password or PIN.
- Use the administrator web interface.

7.34.2.1 Set the log off restriction option in the configuration file

If all the phones in a domain will have one of the last two options above (*Default number used* or *Permitted individual log on*), it is possible to set the parameter **LogOffRestriction** in the configuration file. If the parameter in the configuration file is used the phones will get the directory number that is currently logged on, as the default number.

If the option *Log off allowed* is used and if all the telephones shall be forced to be logged off at a certain time, it is possible to enable a parameter LogOffTime in the configuration file to set the time.

When the parameter *LogOffRestriction* is used in the configuration file, it is not possible to change the log off restriction options locally in each phone. These options are grey in the menu above and the **Select** key is missing.

If the phone is not automatically logged on after the configuration file is read, there are the following possibilities to enter the default directory number and or the password:

- Enter the administrator mode and the **Log on with** menu will appear after a while. Log on with the default number and the password.
- Use the administrator web interface.

7.34.3 Allocation of function keys

Most of the functions allocated to the function keys can be moved or removed, but the Line key(s) are fixed. If the default allocation is to be changed, the system administrator has to modify the configuration file, see description for *CONFIGURATION FILE FOR DBC 42X*.

To enable storing the Dial-by-function key (TNS) numbers in the PBX, 7.42 Dial-by-function keys on page 78.

In the case when the numbers associated with TNS keys are stored in the PBX, special attention is needed to avoid problems that keys are moved upwards or downwards when function keys are added or erased. For more details, see description for *CONFIGURATION FILE FOR DBC 42X* section FUNCTION KEYS.

When the DBC 422 phone is used with MX-ONE TSW or TSE there are no spare function keys available, if the default configuration is used. To make it possible to initiate TNS or MNS numbers the function on at least one of the function keys must first be removed, see description for *CONFIGURATION FILE FOR DBC 42X*.

7.34.4 Change IP settings in DBC 422

It is possible to use the menus in the phone or the web interface to change the IP settings. When using the menus, select the **Settings** menu by using the **Settings** key, enter the administrator mode, press **+** until the **Network** menu appears. Select Network and the list with network settings is shown. Below is an overview of all the **Network** menus:

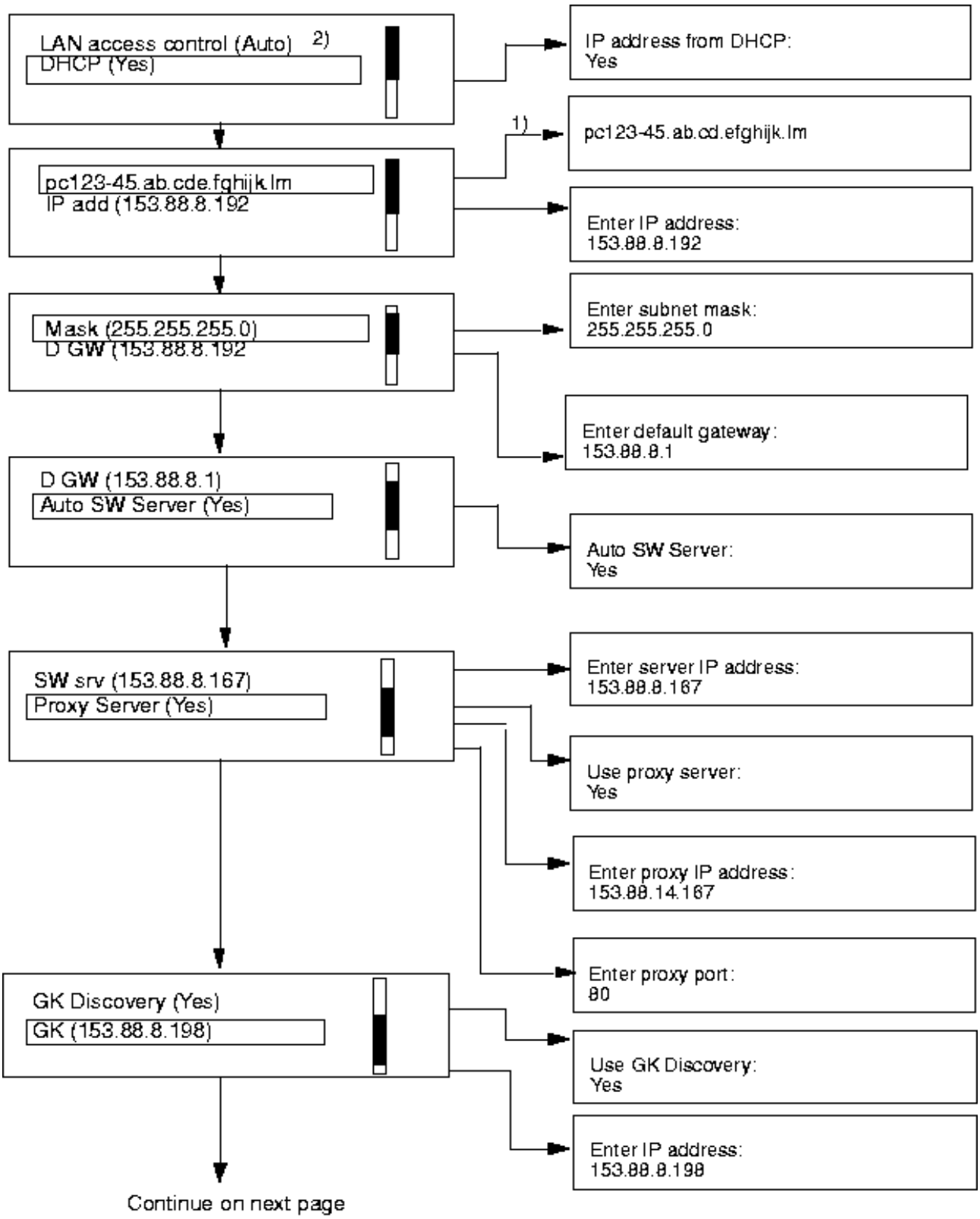


Figure 61: Network settings menus in DBC 422

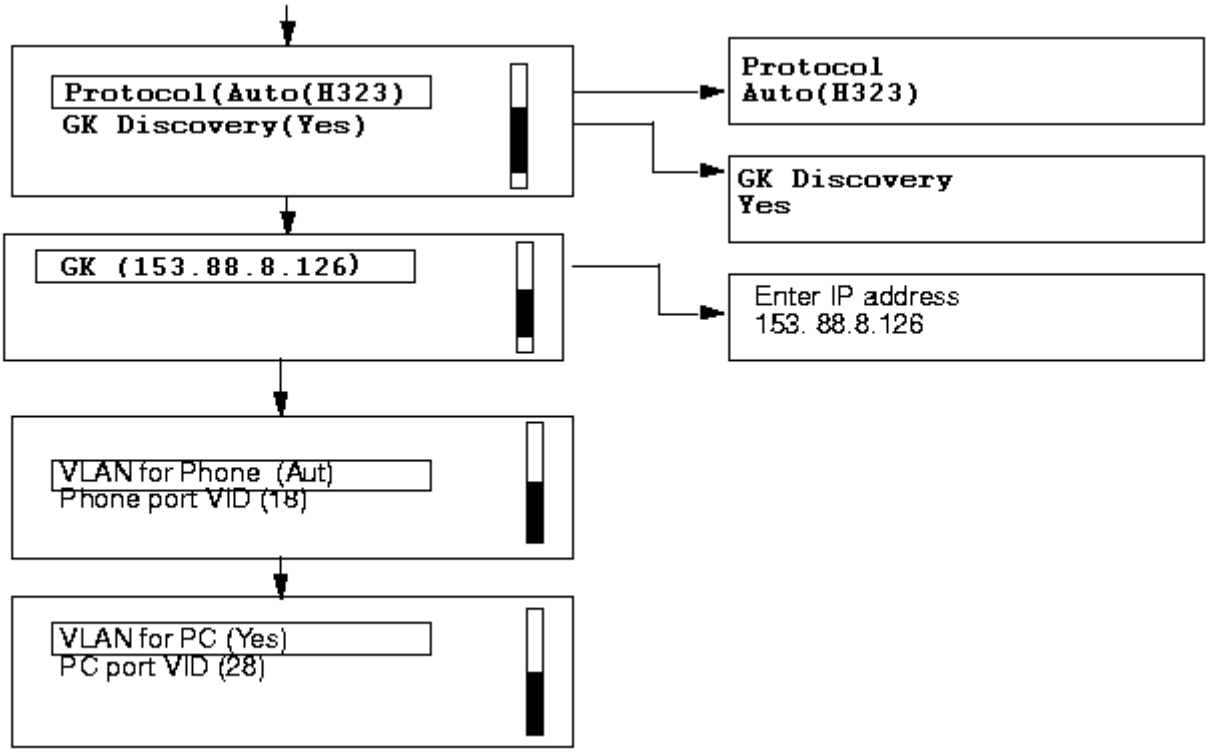


Figure 62: Network settings menus in DBC 422

1) This line is only displayed if a DHCP server is used. It shows the DNS (Domain Name Server) name of the phone and it is used when accessing the phone from a web browser.

The table below explains the abbreviations in the **Network** menus:

Table 6

Display text	**Explanation**
Mask	Subnet mask
D GW	Default gateway
SW srv	Software server
GK	Gatekeeper
VLAN for Phone	The phone port will support VLAN
Phone port VID	The VLAN identity for the phone port
VLAN for PC	The PC port in the phone supports VLAN
PC port VID	The VLAN identity for the PC port

7.34.5 Change IP settings in DBC 425

To change the IP settings it is possible to use the menus in the phone or the web interface. When using the menus, select the **Settings** menu by using the navigation key, enter the administrator mode, press **Down** (F3) until the **Network** menu appears. Select Network and the list with network settings is shown. Below is an overview of all the **Network** menus:

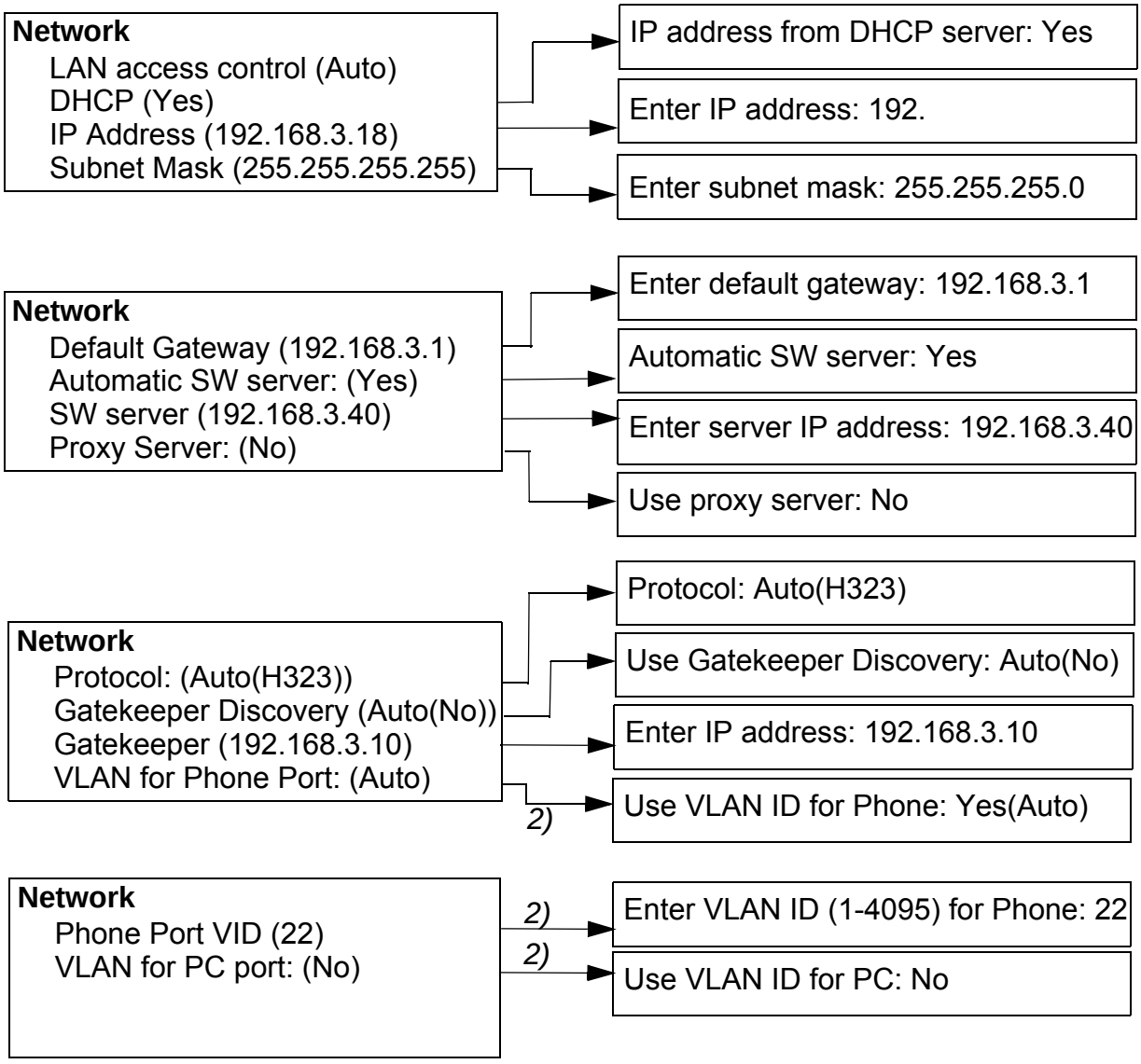


Figure 63: Network setting menus in H.323

2) Changes can only be done in the boot sequence.

7.35 Absence services

(Only DBC 425). The phone has menu support for activation and deactivation of:

- Absence reason (message diversion)
- Follow me
- External follow me
- Profile for personal number

When the user selects an absence service the phone sends the procedure (*n#) to the PBX.

The absence services are defined in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*. If one absence service is not defined in the configuration file, the corresponding menu is not possible to access.

The absence reasons must be synchronized with the absence reasons defined in the PBX and in the support system application (for example, CMG or D.N.A).

7.36 Update of the IP phone software

This section describes the procedure when a new version of the software is to be loaded in the IP phones. The phone will fetch the new version of the software from the SW server, when the phone receives a specific command from the gatekeeper or when the phone is restarted. The procedure is:

- In the product Revision Information (PRI) document for the new version of the application, it is defined if there are new parameters in the configuration file. The PRI is accessible in the same way as a Service Advice:
  - If there are no new parameters in the configuration file: Update the existing configuration file with the new firmware versions.
  - If there are new parameters in the configuration file: Adapt the new configuration file with the existing site dependant parameter values.

  Store the configuration file on the software server, 7.9.3 Directory structure on page 25.
- Store the new bootROM and application software on the SW server, 7.9.3 Directory structure on page 25.

- Perform the update of the phones according to the method for the current system. For MX-ONE see operational directions for *IP EXTENSION*.
- For the phones that are registered in the gatekeeper the following applies:
  - The phones will download the configuration file from the software server every 24th hour to check for new software. This option must be enabled with a parameter in the configuration file, see description of of Configuration file for DBC42x.
  - by a command from the gatekeeper, the telephone can be ordered to load the new software.
- For the phones that are not registered in the gatekeeper the following is valid: The gatekeeper does not know the IP address to these phones. For DBC 42x 01 phones the phones have to be restarted manually. For DBC 42x 02 phones:
  - the phones will once every 24:th hour fetch the configuration file from the SW-server to check if new firmware shall be loaded.
  - the IP Phone Administrator tool can be used to get a list of these phones, 7.13 IP Phone Administrator Tool on page 43. From this tool it is possible to open the Web server interface to the phone, select **Network**, press the key **Apply all settings**. The phone will reboot and update the software.
- To start the update process manually from an IP phone, press the keys **C** (clear key), **mute** and # simultaneously for a second to restart the phone. The update process may take about one minute.
- Alternatively, if the phones are power fed from a power hub, it is possible to update the phone by power off/on the phones centrally.
- Verify that the right version of the software has been loaded by using a print command in the gatekeeper or 7.25 Software version on page 63.

7.37 Backup gatekeeper for branch offices

In a branch office scenario where the IP phones in the branch office are connected to the PBX in the main office, it must be possible to make calls even if the connection to the main office is lost. The solution for this is to use a backup gatekeeper locally in the branch office. When the connection to the main office is lost the IP phones in the branch office automatically register to the backup gatekeeper. When the connection to the

main office works again, the IP phones un-register from the backup gatekeeper and register to the PBX in the main office.

The procedure to get this working in the IP phones is:

- Define in the configuration file, used by the phones in the branch office, the type and the IP address of the backup gatekeeper, see the description for *CONFIGURATION FILE FOR DBC 42X*.
- The frequency of the keep alive check from the phone towards the gatekeeper must be considered. The recommended value is one minute, which means that maximum one minute and 9 seconds (the check is performed 3 times with 3 seconds pause) after the connection to the main office is lost, the phones in the branch office will try to register towards the backup gatekeeper. The drawback of setting the time too short is that the network will be loaded with such a messages. See the data identifier **RRQTtl** in the description for CONFIGURATION FILE FOR DBC 42X.
- The frequency of the routine for discovering when the main office connection is working again is also defined by the data identifier **RRQTtl** , see the description for *CONFIGURATION FILE FOR DBC 42X*. When the main office connection is working, the phone will be registered to the main office gatekeeper.

7.38 Emergency call

There are two cases of emergency calls:

- from an IP phone which is not logged on. See below.
- from an IP phone which is logged on. The call is handled as an ordinary call using the IP extension interface. The sent A-number is the extension number of the logged on user.

For MX-ONE see operational directions for *EMERGENCY CALLS, SOS CALLS*.

An IP phone which is not logged on

The emergency number as well as the IP address and other data for the server which will be used for the call are defined in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

In the configuration file it is also possible to define the A-number to be sent. This should be the A-number associated with the geographical area where the phone is located.

A group of phones that will send different geographical A-numbers compared to another group of phones, must use different configuration files, 7.8 Several configuration files on page 22.

When the emergency call function is enabled in the configuration file, the emergency number is shown in the log on menu. When the user lifts the handset the dial tone is heard although the phone is not logged on. When the user dials the emergency number, the phone uses the IP trunk interface to establish the call. The Setup is sent directly without any admission check.

The A-number defined in the configuration file is sent. The number sent to the public exchange must be within the direct-in-dialling number series, otherwise the public exchange will replace this number.

After the emergency call is terminated the phone returns to the not logged on state. The emergency centre can call back to the terminal although it is logged off.

It is possible to define a first and a second choice for the emergency call server in case of that the first choice fails.

Note: As soon as the emergency number is defined in the configuration file, the log on menu in the phones indicates that it is possible to dial the emergency number. But it is very important that the emergency number is set up in the PBX and tested before it is enabled in the IP phones.

Note: Verify that it is possible for the alarm centre to call back to the number that is sent as the A-number. One possibility is that the number is answered by the PBX operator.

7.39 Monitoring key (MNS key)

It is possible to monitor other extensions from programmable function keys on the IP phone. This function is also called MNS (Multiple represented directory number with dial-by-function key) and is often used in Boss-Secretary applications.

The Monitoring keys are initiated in the PBX. The only changes that can be done by the end-user is the changing of the type of ring signal for the Monitoring key. The only parameter that can be changed in the configuration file is the delay time before the ring signal is generated for the Monitoring keys, see the description for *CONFIGURATION FILE FOR DBC 42X*.

7.40 Call Park Pool

For a detailed description of the Call Park Pool feature in an MX-ONE environment, see operational directions for Call Park Pool.

No configuration in the phone is needed for this feature.

7.41 Automatic Answer

With this feature the call is answered automatically in handsfree mode.

This feature is set by the system administrator. The following options are available:

- With delay, which means that one ring signal is heard before the call is answered
- No delay, the call is answered immediately

To set this feature, go to the **Settings** menu, enter administrator mode and then the **Auto answer** menu.

7.42 Dial-by-function keys

The numbers assigned to the Dial-by-function keys can be stored in the PBX (if the software version of the exchange has this function). This makes it possible for the end-user to bring the numbers when logging on to different phones. To enable this storing of the numbers in the PBX, the parameter **EnablePBXStoring** must be set in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

Note: When the PBX is upgraded from a software version that cannot store the Dial-by-function key data to a software that has this function, the data of the keys will be lost the first time the phone is logged on.

7.43 Missed Call

One of the options in the call list is Missed calls. In the case when a user has several telephones using a feature like parallel ringing or multiple terminals service, a call is marked as a missed call although the call is answered on another telephone.

It is possible to set a timer, meaning the time in seconds in ringing state before an incoming call is regarded as a missed call. The timer is set in the configuration file.

7.44 Operator Media Device (OMD)

The phone can be used in a PBX operator solution based on IP. In this solution CMG’s NOW or D.N.A’s Operator Work Station (OWS) is used

for the call handling and the phone (OMD) is used for the speech. OMD can only be used with MX-ONE.

The interface between the OMD and the gatekeeper is not based on H.323, it uses a proprietary signalling. Depending on that the gatekeeper is not using H.323, it is called the telephony server.

The IP address and port number to the telephony server are defined in the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

To register the OMD to the telephony server, the directory number and password must be entered from the phone. The directory number and password must first be defined in the telephony server.

The phone can play call progress tones but these are generated in the PBX.

The ring signal has two options:

- The PBX sends low ring signal level. The phone generates a low ring signal where the volume cannot be changed.
- The PBX sends high ring signal level: The PBX operator can change the volume of the ring signal with the volume keys.

7.44.1 Set the phone to be used as an OMD

The phone can be used in a PBX operator solution based on IP, 7.44 Operator Media Device (OMD) on page 78.

1) Enter the **Settings** menu.
2) Enter the administrator mode.
3) Go down in the list until the **OMD** menu appears and press **Select** (F4), or the Speaker key for DBC 422.
4) Change to OMD = Yes.
5) Press **Save** (F4), or the Speaker key for DBC 422.

It is also possible to set this data from the administrator web interface.

7.45 Corporate Directory

(Only DBC 425). From the telephone it is possible to search in the CMG directory or in the D.N.A directory.

The telephone sends an http request with the search criteria to the directory server and receives a list with the search result. The answer can be in XML format or in WML format and are decoded by the telephone with

an XML parser or a WAP browser. The user can select the telephone number in the search result and initiate a call via the gatekeeper.

The corporate directory tab **CorpDirectory** is only visible if the IP address to the corporate directory server is enabled in the configuration file of the telephone.

When using the WAP interface and if another directory server shall be used for Corporate Directory, the XML (html) pages must look similar to the pages received from CMG or D.N.A.

To be able to access the directory function some parameters in the configuration file of the telephone have to be set, see description for *Configuration File for DBC 42X* section WAP. In the same section there is also a description of how to convert the number stored in the corporate directory to use it to initiate a call in the PBX. For example if +46 8 56867609 is received from corporate directory, the country code must be removed and a route access code added.

As an alternative to use the WAP browser in the terminal to access the corporate directory, it is possible to use an XML interface. One reason for using the XML interface is that the WAP browser has some limitations when it comes to special characters in some languages. One other reason can be that the GUI is improved in the XML interface.

The XML interface can only be used with the directories in CMG and in D.N.A.

7.45.1 CMG Directory

See *Corporate Directory for IP Phone - Installation and Configuration Guide* in the CMG documentation.

7.45.2 D.N.A. Directory

The IP phone can access the D.N.A. directory through the D.N.A. Mobile Executive (DME).

In D.N.A. Mobile Executive the user agent = DBC42501 has to be defined.

7.46 WAP portal

(Only DBC 425). From the tab **Web** it is possible to access Internet pages adapted for mobile devices and view the content on the phone display. The WAP browser in the phone supports WML, HTML, XHTML basic and XHTML mobile profile.

To make it easy to access such pages, a WAP portal with links to the wanted internet pages can be created. The portal (home page) must be adapted for each customer site.

The address of the WAP portal is defined in the configuration file see the description for *CONFIGURATION FILE FOR DBC 42X* section WAP.

This is an example of how a customer's WAP portal could look:

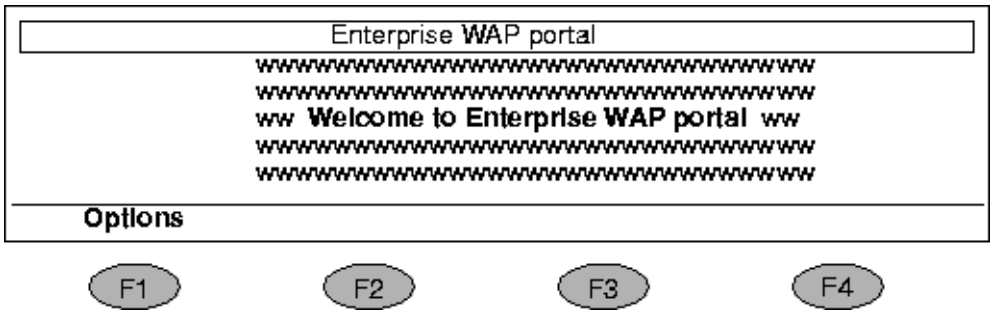


Figure 64:

This menu would be shown for a couple of seconds and then the following menu would appear:

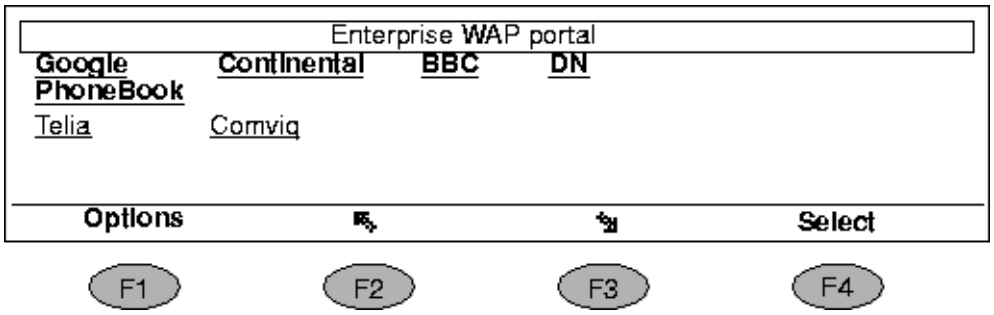


Figure 65:

Information about creating WML pages can be found at: www.w3schools.com/wap/

7.47 Central Storage of the Local Phone Contacts

When an end-user registers on different phones it is possible to bring the Phone Contacts to the phone used at present, by storing the Contacts on a FTP server. Secure FTP (SFTP) can also be used, see section 7.47.4 Secure FTP (SFTP) on page 84.

This is also mandatory when using My Dialog 4000 Contacts, see section 7.48 My Dialog 4000 Contacts on page 85.

Note: DBC 422 can have local contacts but these can only be managed via the web interface.

The Phone Contacts are stored on an FTP server that can be defined in the configuration file.

The stored file will be loaded from the FTP server when the phone is registered to the gatekeeper.

The name of the file consists of the extension number to the user with the file type **.txt** example: 67609.txt

A user identity (user account) has to be created on the FTP server. This user identity and the password are defined in the configuration file for the phone. The default user identity is *Telephone* and the default password is also *Telephone*.

Note: Make sure that the user identity has access rights to read and write the Phone Contacts files on the FTP server.

At registration, the Phone Contacts in the phone will be replaced by the Phone Contacts loaded from the FTP server. After the Phone Contacts has been edited in the display menu and the user has pressed exit, or when the Phone Contacts has been changed in the web interface, the entire Phone Contacts will be stored on the server.

The function can be switched on/off via the configuration file, see the description for *CONFIGURATION FILE FOR DBC 42X*.

The phone has been tested with the FTP servers below.

7.47.1 Windows® IIS

When using Windows® IIS as the FTP server, right-click on **My Computer** , select **Manage** to get to **Computer Management** . First the user identity and password has to be created:

- Expand **Local Users and Groups**
- Right-click on **Users** and add a new user. Enter the same user identity and password as defined in the phone configuration file.
- Disable **user must change password at next log on**. Enable **password never expires**.
- Click on **Create** and the close

Then the directories have to be created:

- Expand **Services and Applications**
- Expand **Internet Information Services**
- Create a directory where to store the files for user specific data. This directory must be the home directory for the user created above.
- Select **Default FTP site**

- Right-click and select **New**
- Select **Virtual Directory**. Name the virtual directory according to the physical directory created above. Enable read and write as access permission.

7.47.2 Fastream (for Windows)

Fastream NETFile FTP / Web server can be downloaded from www.fastream.com. When this FTP server is used, the web server part in Fastream NETFile must be switched off.

This procedure must be followed in the FTP server:

- The account and the password must be created. Use the same values as defined in the configuration file for the phone.
- Select the **Path** tab, create/add a home directory for the account. Select the option **Home Folder**.
- For this home directory, select **File Rights** options **Download**, **Upload** and **Delete**, select **Folder Rights** option **Change**.
- Under the tab **Cache**, disable both **Folder Cache** and **File cache**.

7.47.3 Linux

The recommended FTP servers to use on Linux are **pure-ftpd** or **vsftpd**, which are usually bundled with the used Linux installation distribution. If not they can be downloaded from the following web sites: www.pureftpd.org www.vsftpd.beasts.org

A user account has to be created on the Linux machine where the FTP server will be hosted. The account must be according to the user id and password defined in the configuration file for the phone (the default user identity is *Telephone* with password *Telephone*.

To perform the operations described below, root access is needed.

To create the user account and password it is possible to use any of the GUI tools included in most of the Linux distributions. It is also possible to use the **useradd** and **passwd** commands. Example with the default account and password *Telephone*:

ftp-server: # useradd -g users -m -s /bin/false Telephone ftp-server: # passwd Telephone

Note: The parameter -s /bin/false prevents anyone from using the account to log on to the FTP server using Telnet or SSH (Secure Shell).

To install the FTP server it is possible to use the GUI tools or to use commands. Depending on that the different GUI tools require different

actions, they will not be described here. Please refer to the documentation for the used Linux distribution.

There are two ways that the FTP server can be used either as a stand alone server or as a part of a super server. To keep this section brief there is only a description of how to set up the FTP server as a part of the **xinetd** super server.

7.47.3.1 Pure-ftpd

This FTP server is set up by use the command line option. Open the file /etc/xinetd.conf in an editor and add the following:

service ftp { socket_type = stream protocol = tcp wait = no user = root server = /usr/sbin/pure-ftpd server-args = -A -E -H -i -u 500 -c 1000}

The explanation of the arguments can be found in the documentation for the FTP server.

As an alternative this can be set in the file /etc/xinetd.d/pure-ftpd instead.

Restart the super server with the command:

ftp-server: # killall -USR2 xinetd

7.47.3.2 vsftpd

This FTP server is set up by using a configuration file. The options listed below are a subset of all available options.

Open the file /etc/vsftpd.conf and edit the following options:

write_enable=YES ls_recurse_enable=YES local_enable=YES chroot_local_user=YES anonymous_enable=NO connect_from_port_20=YES pam_service_name=vsftpd

All the remaining settings can be disabled by creating a comment (# character).

Then open the file /etc/xinetd.conf in an editor and add:

service ftp { socket_type = stream protocol = tcp wait = no user = root server = /usr/sbin/vsftpd log_on_failure += USERID }

As an alternative this can be put in the file /etc/xinetd.d/vsftpd instead.

Restart the super server with the command:

ftp-server: # killall -USR2 xinetd

7.47.4 Secure FTP (SFTP)

The user name and password to log on to the SFTP (or FTP) server is set in the configuration file of the telephone. The default user name is Telephone.

In a SFTP server running on Linux the following must be considered:

The **.txt** files with contacts are stored under the folder: **/home/Telephone**, when using the default user name. These folders must be enabled for read and write access.

In the Linux environment the following parameters have to be set in the file: /etc/ssh/sshd_config

PasswordAuthentication yes
AllowUsers Telephone

If the user name and password to the SFTP server shall be something else than default, the configuration file of the phone and /etc/ssh/sshd_config must be updated.

7.48 My Dialog 4000 Contacts

My Dialog 4000 Contacts is an application that makes it possible for the end-user to merge the contents in Microsoft® Outlook Contacts to the existing Phone Contacts in the IP phone. This application updates the Phone Contacts on the FTP server, and after that the phone is updated with the new Contacts in a file from the FTP server. This means that the Phone Contacts must be stored centrally on the FTP server, 7.47 Central Storage of the Local Phone Contacts on page 81.

It is important how the end-user stores the external phone numbers in Microsoft® Outlook Contacts. If the external destination code starts with 0 and the area code also starts with 0, the phone cannot distinguish those numbers. If an external number in Outlook is stored with the external access code, it will **not** work. The recommendation is to store the complete external numbers in Outlook, such as +country code, area code, phone number. Example: 46 is the country code, 08 is the area code and 7190000 is the phone number, store +4687190000.

My Dialog 4000 Contacts program shall be stored on a web server and the end-user shall get a link to a web page from which this application can be downloaded to his/her PC. Java run time library 5.0 or higher is needed on the end-used PC (can be downloaded from the Sun® home page).

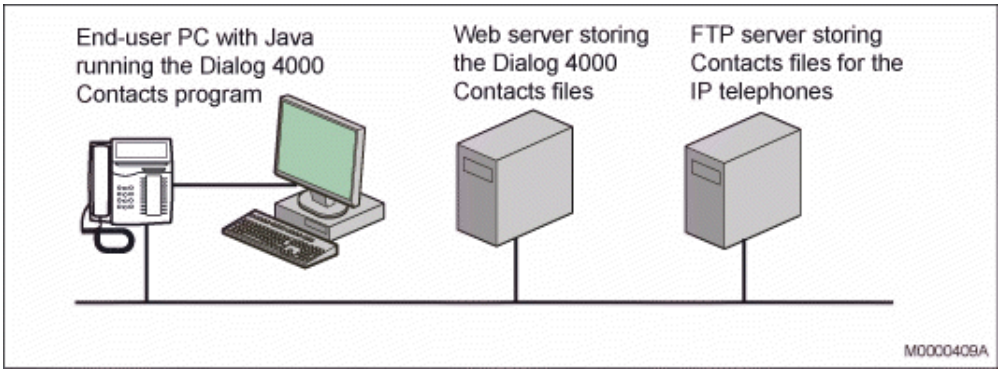


Figure 66:Components in My Dialog 4000 Contacts

For a description of how to use the application **My Dialog 4000 Contacts,** see the directions for use for each platform.

7.48.1 Installation of My Dialog 4000 Contacts

Deliverable

This application is delivered as a zip-file containing:

- A folder **My Dialog 4000 Contacts** containing the program files to install and a read me file.
- A sample web page, **dialog4000webstart.htm**, containing the code to include on a company web page for the user to download the application.
- An example of a company web page **index.html**

Installation of the program files:

- Unzip the zip file to a temporary location
- Copy the folder **My Dialog 4000 Contacts** to a location on a web server that the end-users can access.

The end-user starts the downloading of My Dialog 4000 Contacts from a web page (this file is called **index.htm**). This web page has a link to **dialog4000webstart.htm** and shall also contain information and instructions to the end-user. The included web page is an example, which can be changed to a layout according to the customer's need.

The **dialog4000webstart.htm** file contains sample code to enable download of the **dialog4000.jnlp** file. In **dialog4000webstart.htm** the reference to the appropriate URL where the application is stored, has to be set.

When using an IIS server as the web server, the file type **.properties** must be enabled in a similar way as described above, 7.9.2 HTTP servers on page 24.

For more details bout the installation, see the read me file included in the zip file.

Configuration

The file **dialog4000.jnlp** contains the configuration parameters. The file can be updated with any text editor. The parameters are described in the read me file.

7.49 IP Voice Recording

It is possible to record voice calls to a central recording equipment. The phones that shall have recording are monitored via the CSTA interface and this means that an Application Link or an Open Application Server (OAS) must be used to provide the CTI interface to the recording system. The call events and the IP address to the phones to be monitored are sent over the CSTA interface.

For more information about the recording solution for MX-ONE Telephony Server see *Description for Voice Recording* and the *Interface Description for VoIP Recording Interface.*

The signalling between the recording system and the IP phones is based on SIP, although the phone is using H.323. The recording system sends an INVITE message to the phone to inform about the IP address to where the voice packets shall be sent. A SIP ACK message orders the phone to start forwarding the received and transmitted RTP streams to the logger.

There are the following options:

1. Total recording: all calls to the monitored calls are recorded
2. Record on demand: the user can start and stop the recording by pressing the recording key.

Note: It is only possible to record IP phones. No other types of phones shall be monitored.

The voice stream is sent un-encrypted to the recording equipment, if the original call is without encryption. If the call is encrypted, the telephone forwards an encrypted voice stream to the recorder. In this case the encryption keys are sent via the CSTA interface to the recording equipment.

There are a number of parameters in the configuration file for voice recording, see *Configuration File for DBC 42x*.

The shortcut key for recording on the phone is initiated from the PBX.

Total recording

The recording key is lit as soon as the telephone forwards the RTP stream to the recording system.

Record on demand

The LED on the recording key on the phone is lit when the end-user has ordered the call to be recorded by pressing the key and the phone has got a confirmation from the recording system.

The URL that the telephone sends to the recording system, when the user presses the recording key, can be defined in the configuration file.

7.50 Wall mounting of the IP phone

The phone can be wall mounted, useful for instance in conference rooms or in public areas. Use the wall mounting kit SXK 106 2049/1, which consists of the spacer SXA 112 4753/1 and two screws.

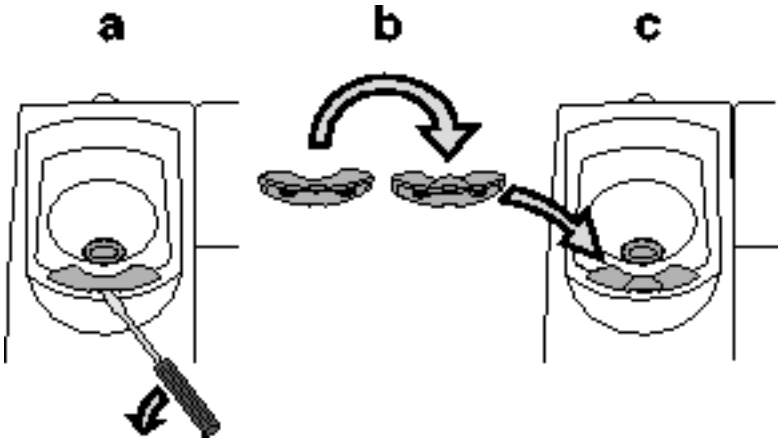


Figure 67:

- Use a screwdriver to remove the handset hook.
- Turn the hook upside down and put it back.

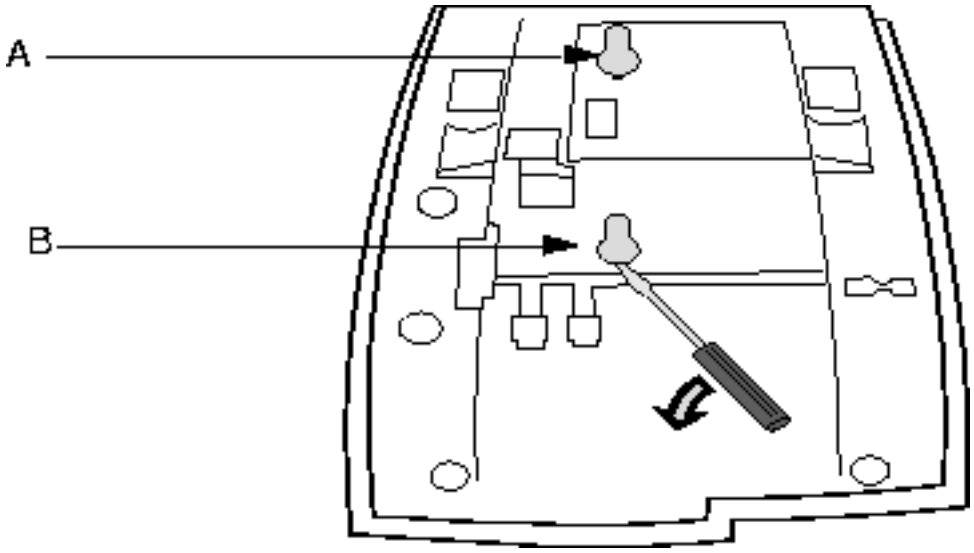
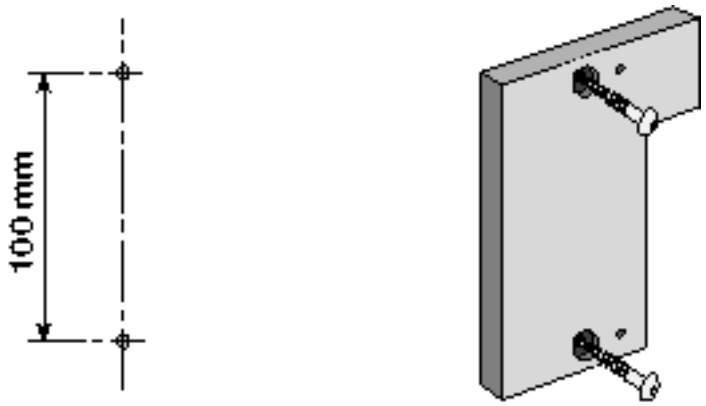


Figure 68:

- Use a screwdriver to remove the two (A and B) plastic covers
- Drill two wall holes and mount the spacer SXA 112 4753/1 on the wall. These holes must be 15 mm more to the left than the centre of the phone, otherwise it will land up 15 mm to the right, after the mounting.



Hole dimensions depending on type of wall.
Wall screws are not supported (∅ max. 5 mm).

Figure 69:

- Fasten supplied screws to the spacer.

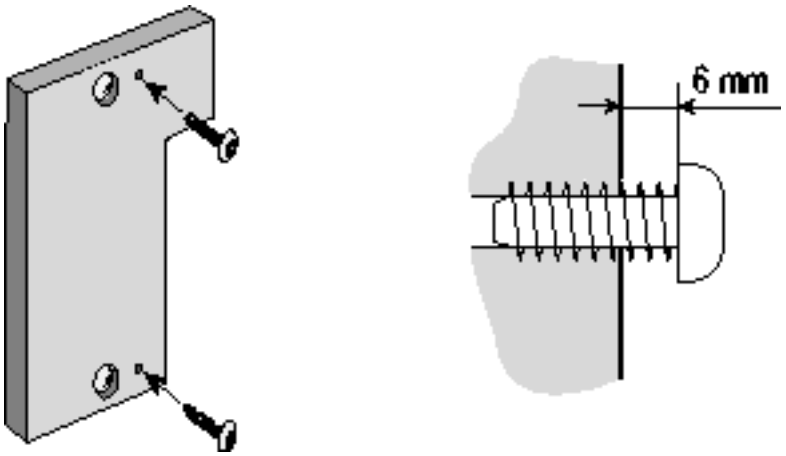


Figure 70:

- Hook the phone on the spacer screws

Measures for a locally manufactured spacer

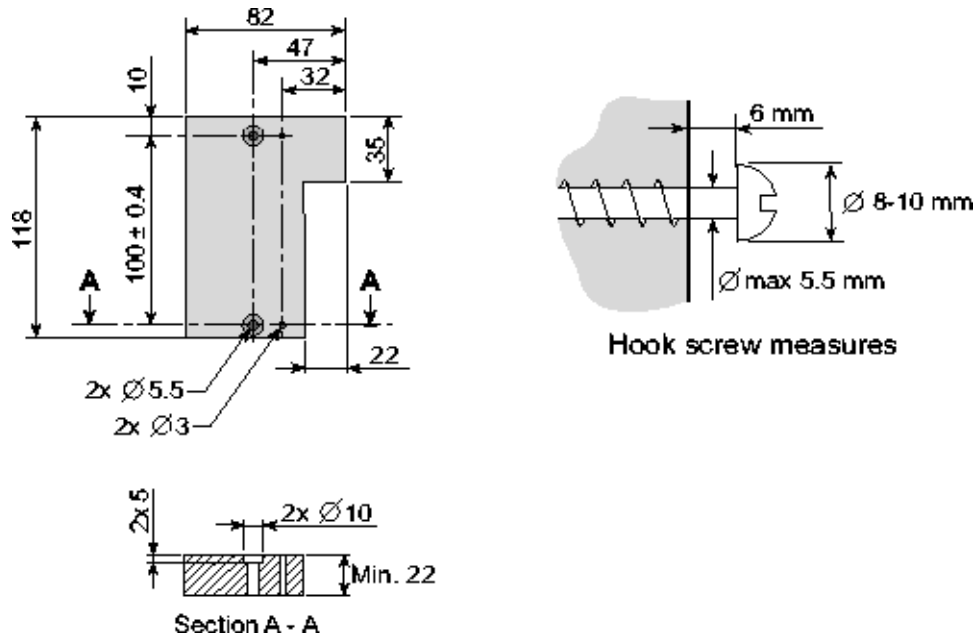


Figure 71:

7.51 Post installation measures

Check that it is possible to log on the phone to the system.

Verify that internal and external calls can be established from and to the phone.

8 Installation of extra key panel DBY 419 01

DBC 422 can have maximum one extra key panel.

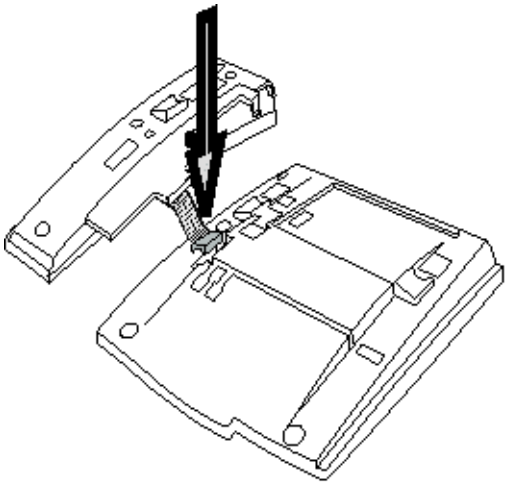
DBC 425 can have maximum four extra key panels.

The keys can be used as Dial-by-function keys and Monitoring keys. It is also possible to assign functions (call back, free on second etc.) on these keys but it is not recommended because when a user logs on to a phone without the extra key panel, this function will not be available.

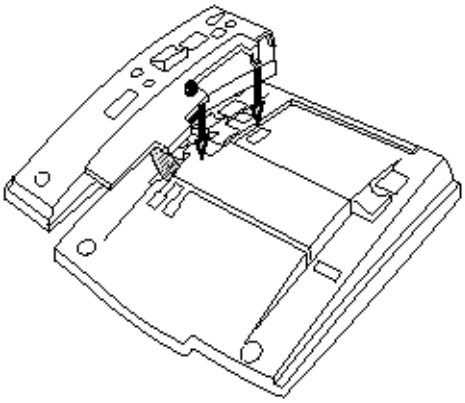
8.1 Installation of extra key panel DBY 419 01

Make necessary arrangements to avoid electrostatic discharges.

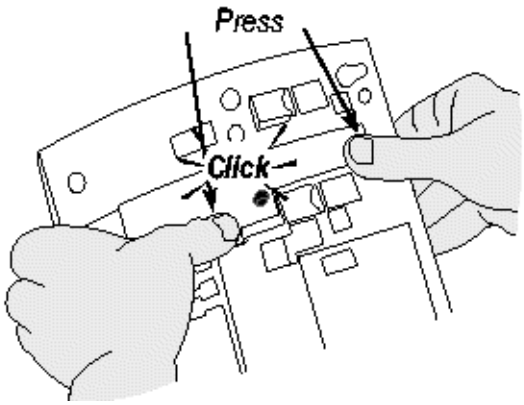
- Disconnect the power from the phone.
- Remove the feet.



Fasten the connector in the bottom of the hole using a finger or a blunt tool. Make sure that the connector fits before pressing it down



Attach/snap the key panel unit to the telephone



Press down the key panel unit

Figure 72:Attaching the DBY 419 01 key panel unit to the phone

Mount the proper quantity of feet in recommended positions, 73 Feet positions with 1 to 4 pcs DBY 419 01 connected (rear view) on page 93.

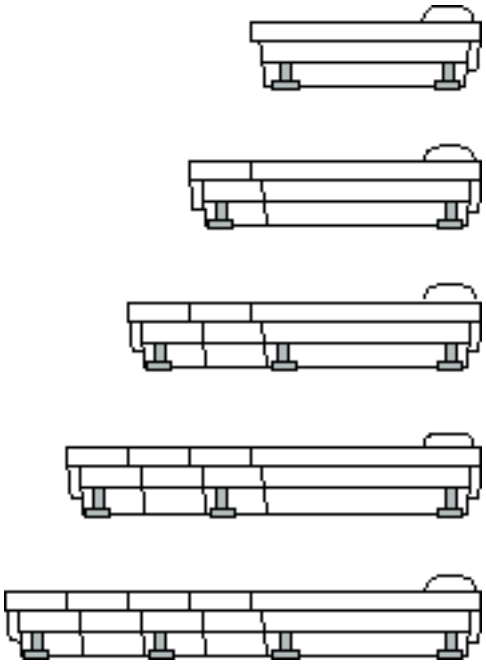


Figure 73:Feet positions with 1 to 4 pcs DBY 419 01 connected (rear view)

8.2 Post installation measures after mounting of the key panel unit

Connect the power to the phone and connect the phone to the LAN. Program one function key per key panel. Establish a call using the key on each DBY 419 01.

9 Installation of option unit DBY 420 01

Make necessary arrangements to avoid electrostatic discharges:

- Log off the terminal and disconnect the power from the phone set.
- Mount the pin connector to the option unit, 74 Mounting of pin connector on page 94. Avoid if possible to touch the contacts.
- Connect the option unit to the phone, 75 Mounting of Option unit to the phone on page 95 and also 76 Press down the Option unit to secure good contact. on page 95
- Secure the option unit with the three screws (Screwdriver for Torx No. T8).
- Connect the power to the phone and when the phone has started, log on the terminal.

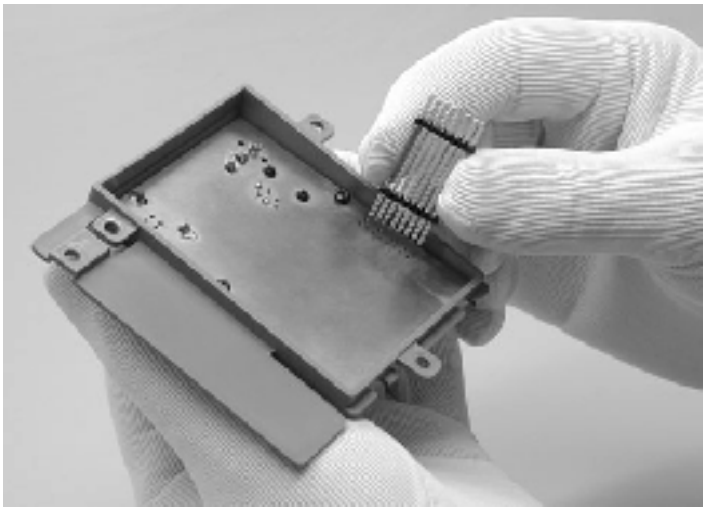


Figure 74:Mounting of pin connector

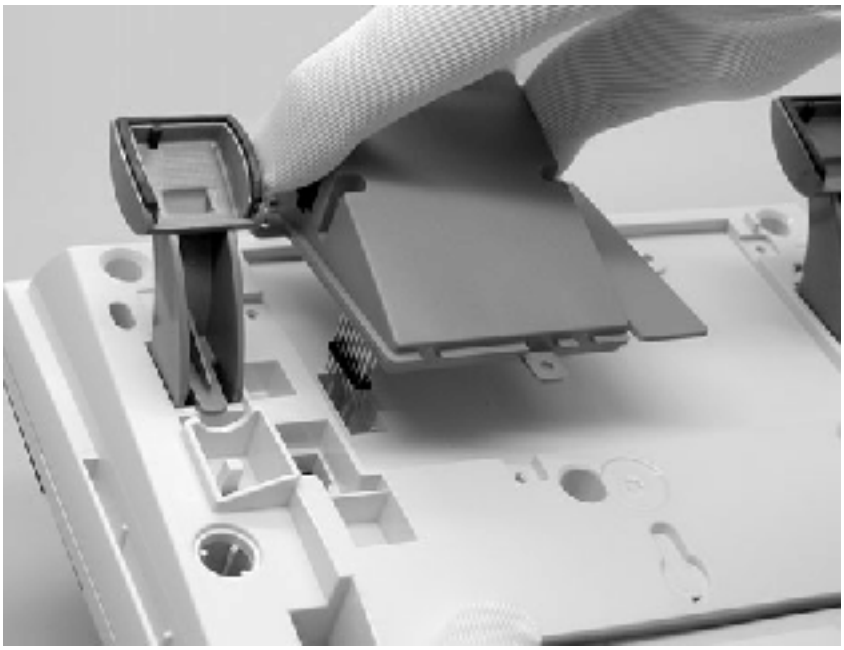


Figure 75:Mounting of Option unit to the phone

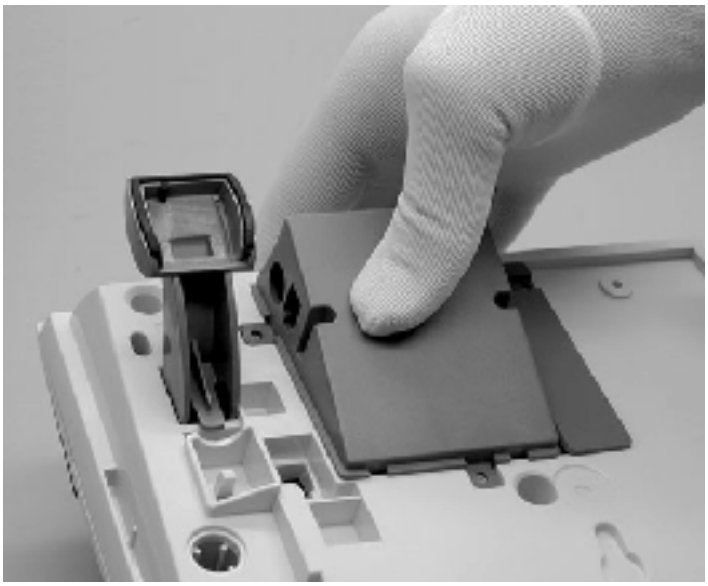


Figure 76:Press down the Option unit to secure good contact.

9.1 Extra bell/ Busy signal

For pin outlets of the external functions, 77 External cable connection on page 96

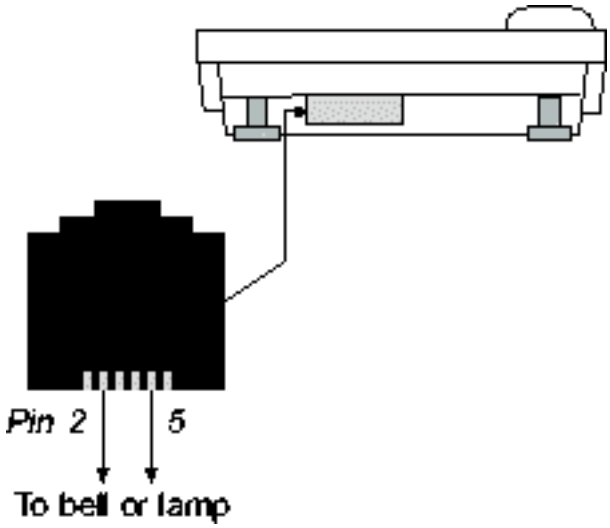


Figure 77:External cable connection

There are three alternatives using this function. The function is selected by the menu: **Settings - Option Unit**.

Extra bell:
The extra bell (or lamp) is activated in parallel with the ring signal. It is possible to define via the configuration file if the bell or lamp shall be activated when a call is received on a MNS key.

Busy signal:
The busy signal is activated in off-hook mode. The function can be used to control a Do-not-disturb lamp at the door.

Combined Extra bell and Busy signal:
Activated in parallel with the ring signals and steady active in off-hook mode. This indication can be used for lamp indication in e.g. office landscapes.

A free on second call does not activate the extra bell function.

When the function is active, the circuit is closed via an opto relay, which is used to separate the external device electrically from the phone. Maximum load on the external device is: 1 A resistive or 0.3 A capacitive or 0.3 A inductive load at maximum 24 V AC or 48 V DC. An external over-voltage protection is recommended.

9.2 Post installation measures after mounting the option unit

Connect the power to the phone and when the phone has started, log on to the gatekeeper. Check that a call can be established and check that the installed extra equipment works.